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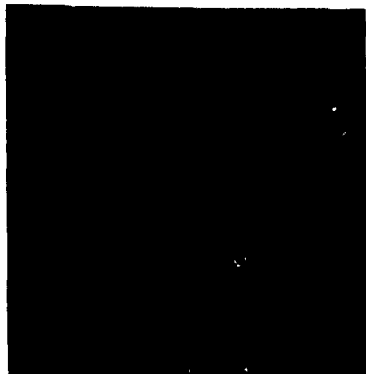


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FINAL REPORT
PASSIVE SPHERICAL SATELLITE
COMMUNICATIONS STUDY

Prepared by the Staff of the
Communications Department

ITT *Federal*
LABORATORIES

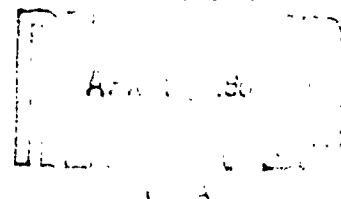
A DIVISION OF INTERNATIONAL TELEPHONE AND TELEGRAPH CORPORATION

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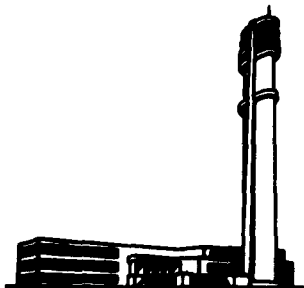
Prepared for
Rome Air Development Center
Air Research and Development Command
United States Air Force
Griffiss Air Force Base, New York

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United States Air Force
Griffiss Air Force Base, New York

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MARCH 1962

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FOREWORD

The following members of the staff of the Communications Laboratory at ITT Federal Laboratories, Nutley, New Jersey, participated in the study program and in the preparation of this Final Technical Report:

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A B S T R A C T

The terminal equipment specifications presented in this report provide the traffic handling capabilities for ground stations operating within a proposed passive satellite communications network. These stations will vary in size, depending upon the amount of traffic flow and the number of links served, but may be built up from a set of basic storage, processing, and switching components.

The nature of the transmission medium poses several problems not found in contemporary communications networks. These problems include: a variable length transmission path, an instant (or interval) of transfer of communications from a setting satellite to a rising satellite, and the handling of traffic in the several forms of voice, graphics, teletype, and data.

It is proposed that all traffic be handled in digital form once it enters the system and that transmission on each link be in the form of a time-division multiplexed serial binary stream. To correct for the variable length transmission path, the bit train is reclocked at the receiving end of each link. At the ground station, traffic is separated into real-time (voice and graphics) and store-and-forward (data and teletype). The two forms are processed through separate switching centers and are then recombined for transmission on the next link.

Equipment is specified which is capable of performing these operations. Cost estimates are given as a function of traffic load.

Final Report

**PASSIVE SPHERICAL SATELLITE
COMMUNICATIONS STUDY**

C O N T E N T S

	<u>Page No.</u>
FOREWORD.	ii
ABSTRACT.	iii
ILLUSTRATIONS	vi
1. INTRODUCTION.	1
2. TRAFFIC STUDY	3
2.1 General Subscriber Characteristics	3
2.2 Reference Network.	3
2.3 Circuit Reliability.	6
2.4 Comneeds on Links.	7
2.5 Precedence	9
2.6 Traffic Statistics	10
2.7 Alternate Solutions.	11
2.8 Jamming Environment.	12
2.9 Operational Procedures	12A
2.9.1 General	12A
2.9.2 System Monitoring	12A
2.9.3 Routing Procedures.	12A
2.9.4 Alternate Routing	19
3. TERMINAL EQUIPMENT REQUIREMENT.	22
3.1 Philosophy of Design	22
3.2 Satellite Factors Influencing Terminal Equipment . . .	24
3.2.1 Oribts and Coverage	24
3.2.2 Satellite Tracking and Transfer	24
3.3 Types of Signals	29
3.3.1 Separation of Traffic	29
3.3.2 Multiplexing.	30
3.3.3 Composition of the Bit Stream	31

C O N T E N T S (cont'd)

	<u>Page No.</u>
3.4 Real-Time Traffic Handling.	33
3.4.1 State-of-the-Art	33
3.4.2 Reference Network Real-Time Center	38
3.4.3 Handling of a Typical Call	39
3.5 Store-and-Forward Traffic Handling.	43
3.5.1 State-of-the-Art	43
3.5.2 Reference Network Message Center	46
3.5.3 Handling of Typical Message.	50
3.6 Equipment Specifications.	52
3.6.1 Storage.	52
3.6.2 Switching.	58
3.6.3 Multiplexing and Demultiplexing.	60
3.6.4 Reclocking	61
3.6.5 Code Conversion.	64
3.6.6 Analog-Digital Conversion.	64
3.6.7 Message Processing	65A
3.7 Equipment Pricing	66
3.7.1 Pricing Basis.	66
3.7.2 Pricing List	68
4. CONCLUSIONS.	71
5. RECOMMENDATIONS FOR FUTURE WORK.	72
6. REFERENCES	73
APPENDIX I - Some Considerations for a Future Ground-Satellite - Ground Communication System	I-1
APPENDIX II - Single Bit Stream Link vs. Two Bit Streams/Link	II-1
APPENDIX III - Optimum PCM Techniques for Reference System . .	III-1
APPENDIX IV - Error Detection and Correction	IV-1
APPENDIX V - Problems Arising from a Variable Time Delay Inserted in a Transmission Path	V-1

ILLUSTRATIONS

<u>Figure No.</u>	<u>Title</u>	<u>Follows Page No.</u>
1	Passive Spherical Satellite Communications Network.	5
2	Probability of Delay > 6 Seconds as a Function of Comneeds Granted Initial Trunk Loading a/c = 0.25	9
3	Alternate Route Networks.	20
4	Alternate Routing Plan.	20
5	Satellite Course Through Joint Coverage Zone. .	26
6	Transmission Paths Without Point of Equal Length	26
7	Transmission Paths With Point of Equal Length	26
8	Joint Coverage Zone Showing Intersections With Ellipsoids	26
9	Joint Coverage Zone for Two Stations.	26
10	Composition of Sample Bit Stream.	32
11	Typical Real-Time Traffic Functional Diagram for Reference Network Station.	39
12	Typical Store-and-Forward Traffic Functional Diagram for Reference Network Station.	46
13	Simplified Switching Array.	58
14	Typical Reed Relay Switching Center	59
15	Reclocking Block Diagram.	61
16	Fielddata Standard Code Table.	64
17	Voice Differential Encoder.	65A
18	Facsimile PCM Encoder	65A

1. INTRODUCTION

This report is the result of a study program which has investigated the problems and requirements associated with the terminal equipment to be used in a proposed passive spherical satellite communications network. The project has been a continuation of a study² conducted during 1960. This traffic study, together with a study of the orbital requirements made by another contractor¹³, specified the reference network which was used as a basis for the present investigation.

Two major areas of interest determine the requirements for terminal equipment within any communications system, the characteristics of the traffic to be handled and the characteristics of the transmission medium. Each area encompasses many individual problems. It is the purpose of this report to discuss these problems, to consider the merits and faults of various methods of accomplishing the desired objectives, and to recommend solutions leading to the specification of particular types of equipment.

In the performance of the study many questions regarding satellite availability, coverage times, and operation developed. To a large extent, these were answered by other participants in the Passive Satellite System study. In some areas, where the complete solutions to these problems would have exceeded the scope of the cognizant studies, general assumptions have been made to permit the continuation of the subject effort. In particular:

- Preliminary calculations of coverage of all B and C stations indicate no outage will occur, hence no buffer storage has been assumed for this situation. Calculations for the extremities of the network indicate outages of approximately

15 minutes duration occurring three times per day for the most northerly stations, hence storage to accommodate circuit outage for this case has been assumed.

- Message statistics on the time distribution density of traffic flow obtained has been inconclusive. For this reason the total messages derived earlier in the study have been assumed to be randomly distributed in time.
- At the writing of this report several methods of antenna transfer from satellite to satellite have been developed. Terminal requirements for these are discussed.
- Facsimile statistics have not been available; therefore, graphic transmission has been assumed to require a real-time transmission channel of 9600-bit capacity.

2. TRAFFIC STUDY

2.1 General Subscriber Characteristics

The nature of the terminal equipment required for the proposed satellite communications network is determined by the characteristics of the inputs to the network and by the capabilities and limitations of the transmission medium. Since the reference network is intended as an Air Force communication system, inputs originate from Air Force Commands and consist of voice, data, teletype, and graphics traffic. The organization of the Air Force is such that the bulk of all communications occurs within each Command and may be generally classed as either operational or administrative in nature. Therefore, it appears that the mission to be accomplished by a particular Air Force Command will establish certain traffic patterns relating to volume, message rates, precedence, reliability of communication, and traffic distribution as a function of time. An estimated volume of traffic to be handled by the reference network was obtained in the earlier portion of this study from an analysis of U.S. Air Force communication requirements. Statistics of actual traffic flow have been analyzed to gain an insight into the expected characteristics of traffic to be handled by the reference network regarding precedence, message rates and length, and time density. The mission of each command concerned has been analyzed to provide an estimate of reliability required. All these factors have been considered in preparation of the proposed system.

2.2 Reference Network

The determination of the network configuration was made during the earlier portion of this study.² For continuity, the manner in which this was accomplished will be reviewed.

A large number of statistics concerning the communication requirements of more than 1,100 Air Force terminals was collected, collated, and tabulated. Requirements of less than 500 miles, which could more efficiently use landlines or other conventional means, were eliminated. Requirements entirely within the continental U.S. were also eliminated because of the elaborate communications systems that already exist.

The initial step in the construction of the network was the establishment of terminal relay points required to service the needs of approximately 500 Air Force stations remaining after the classification described above. Once the location of all participating Air Force bases had been determined, the communication complex was designed so that, insofar as possible, a minimum distance separated all stations from their respective terminals. The resulting information sources and sinks were connected via a tree-type network which was compatible with realizable orbit configurations. The network consisted of a main east-west trunk paralleling approximately 40 degrees north. The nodes of this network represented the terminals to be studied. These terminal stations were then grouped into the three main categories: A, B, and C.

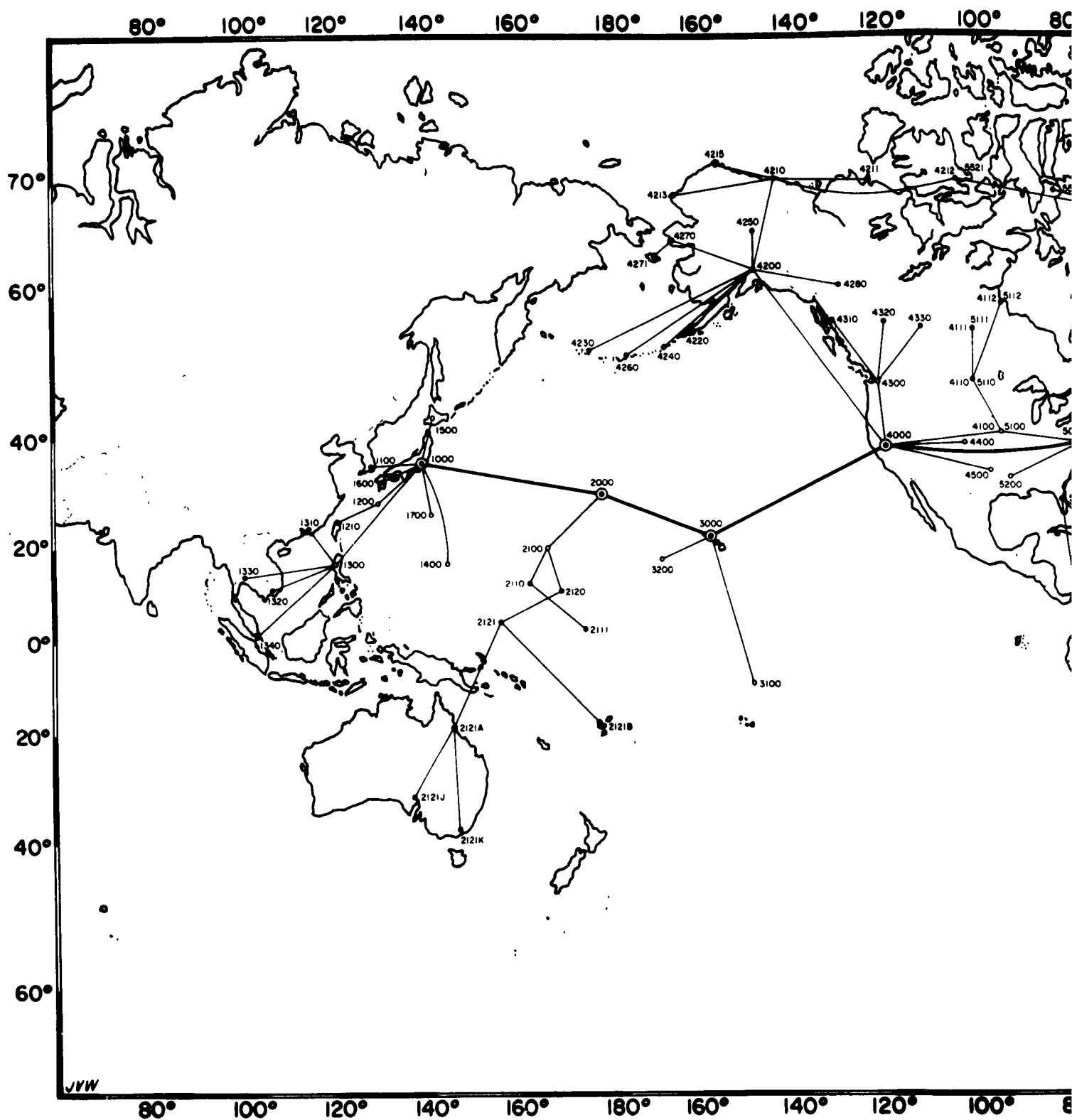
- "A" stations, or primary trunk stations, were selected on the basis of a compromise between the expected location of Aircom switching centers, traffic loads, and the coverage limitations of the satellite orbits. They are located at approximately equal intervals around the earth's circumference, between 20 degrees and 41 degrees north latitude. They act as relay points for primary trunk traffic and B-to-B station traffic.

- "B" stations were chosen as secondary relay points located within satellite range of an "A" station and serving as centers to funnel information to the "A" station for further transmission.
- "C" stations are tertiary relay and collection points clustered around a "B" station.

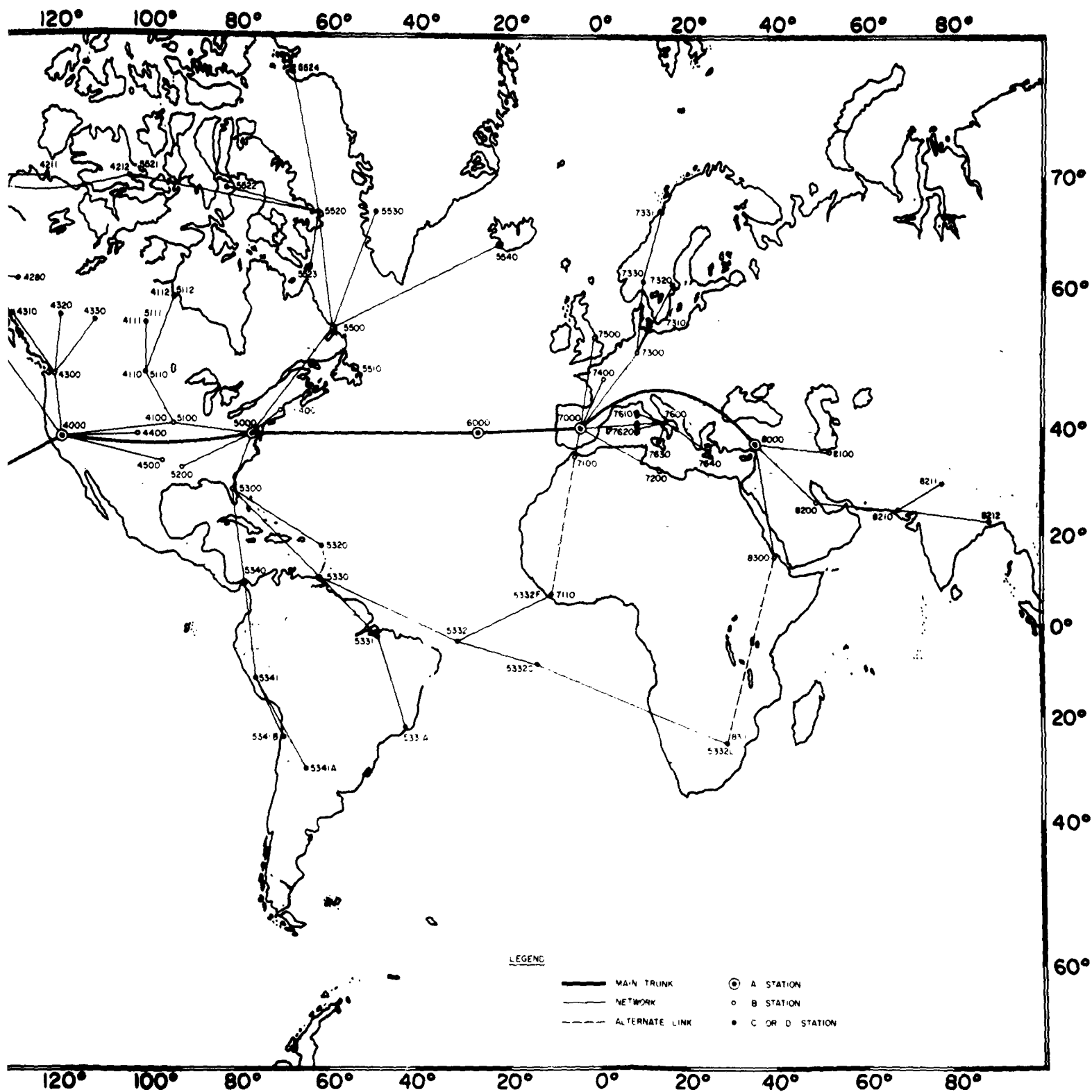
All communication between "A", "B", and "C" stations is via satellites. The three types of stations also act as collection points for reference network inputs originating at Air Force centers within their vicinity. Conventional means of transmission are used in most of these cases to connect with the satellite network.

The locations of the "A", "B", and "C" stations selected for the reference network are shown in figure 1. The "A" stations have thousands designations; the "B" stations, hundreds; and the "C" stations, tens. In a few instances, Air Force centers feeding "C" stations are unable to do so by landline or similar conventional method. These sites, designated by units digits in figure 1, must use satellite links to reach their associated "C" station.

The "A", "B", and "C" stations could be connected in several different ways to form a communications network. The first possibility is to use the trunk or chain-of-command approach. This system resembles a tree with a main trunk and numerous branches. One branch is connected to another through the trunk. The second possibility is the grid system in which each station communicates directly with all other stations within satellite range. Other possibilities consist of various combinations of these two systems. The grid system provides the most direct and rapid



1



2

Figure 1. Passive Spherical Satellite Communications Network

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transmission path. It also reduces the traffic load at relay points. The trunk system follows the route of most of the reference network traffic since military communications generally follow the chain-of-command. It also drastically limits the number of other stations with which each station must be in direct contact.

This limitation results in a considerable saving in terminal equipment since, in a grid system, each link requires additional switching, multiplexing, transmitting, and receiving equipment. It is concluded that with the low altitude orbit and comneeds specified, the trunk system is most suitable for the reference network. The connections between stations follow this system.

2.3 Circuit Reliability

Three grades of circuit reliability have been defined in RADC-TN-61-36A² (Volume I, Circuit Requirements). As indicated, the three grades are:

Reliability A - Extreme reliability, requiring full period alternate route. Maximum permissible delay time, 30 seconds.

Reliability B - High reliability; alternate route can be established by automatic switching. Maximum permissible delay time, 5 minutes.

Reliability C - Normal reliability; alternate route can be established by patch. Maximum permissible delay time, 1 hour.

Since military communications may be generally classed as either operational (high priority traffic) or administrative (low priority traffic), it seems reasonable to consider circuits of A and B reliability as operational and C reliability circuits as administrative channels. Percentage estimates based on command missions as analyzed may be made for each of the major Air Force commands, although these figures will vary for different military or political situations. The table following is an indication of the relative needs for circuit reliability.

	<u>A and B</u> <u>Reliability</u>	<u>C</u> <u>Reliability</u>
AAC	70	30
AFCS	20	80
ADC (NORAD)	70	30
AMC	50	50
MATS	50	50
SAC	60	40
TAC	50	50
USAFE	25	75
USAFSS	90	10

2.4 Comneeds on Links

The basic unit of measure employed in the determination of the reference network was the "Comneed." This is defined as the basic need to communicate, or transfer information, from one location or user to another location or user. Each comneed is fully described by the type of service (voice, graphics, teletype, data), locations and/or users, security and reliability requirements, traffic or usage, and time frame. It does not define the manner in which it can be fulfilled, whether or not it is satisfied by existing facilities, the number of hours per day it exists, or the type of transmission media, routing and switching to be used. Although the comneeds do not provide fully adequate information upon which to base

the complete design of a communications network, they have been used as a guide for the reference network since no more detailed data was available.

The comneeds considered in establishing the reference network represent only about 42 per cent of the total Air Force comneeds. The remainder are for distances less than 500 miles or entirely within the continental U.S., assumed to be provided by conventional communication means. Certain exceptions exist where conventional communications are not adequate to satisfy the needs of shorthaul requirements.

If each comneed is assigned a separate channel, this is known as a fully allocated network. If several comneeds are combined into a single channel, the network is classified as common user. Probably the most important compromise to be made in the entire reference network concerns the number of comneeds per channel. From the viewpoint of speed and availability of communications the fully allocated network is desirable. However, this requires such an extensive number of channels as to be economically unfeasible.

To reduce the comneed data to common user trunk requirements a mathematical model was set up based on assumptions of traffic flow statistics and knowledge of the over-all characteristics of the requirements data. The model is defined by the following:

- (a) Model link consists of five hops. All messages traverse the complete link.
- (b) All messages are A reliability; i.e., maximum permissible delay, 30 seconds.
- (c) Delay is equally distributed; i.e., 6 seconds/hop.
- (d) Probability of success is 0.99.

- (e) Average message length is 3 minutes; average occurrence, one each 12 minutes.

Based on this model and standard telephone practice analysis the trunk requirements were determined in terms of comneeds granted per comneeds requested using curves on figure 2. (a) is justified by the characteristics of the comneed data. (c) and (d) are reasonable assumptions. (a) is valid based on analysis of actual traffic flow as discussed later in this report. Because of the assumption in (b) the final trunk requirements result in a maximum efficiency system. In theory all messages handled meet the delay criteria of A reliability and no system discipline (precedence) is required. However, since the mathematical development has its limitations and 100 per cent probability of success cannot be assured provisions for precedence have been included.

2.5 Precedence

Military systems have used six levels of precedence with transmission requirements ranging from immediate down to twelve hours or more. Recently, automatic message handling systems have been developed which accept all six levels but classify them into either high or low precedence categories. This method is now practical because of the increased speeds of transmission and handling in the new systems.

The decision as to the number of levels of precedence which are required for the reference system is based on several factors. If the analysis of comneed reduction to common user trunks as described above is taken at face value, no precedence system is required. This is because no method of system discipline was taken into consideration in the analysis. However, in practice, the conclusions reached may not be entirely valid

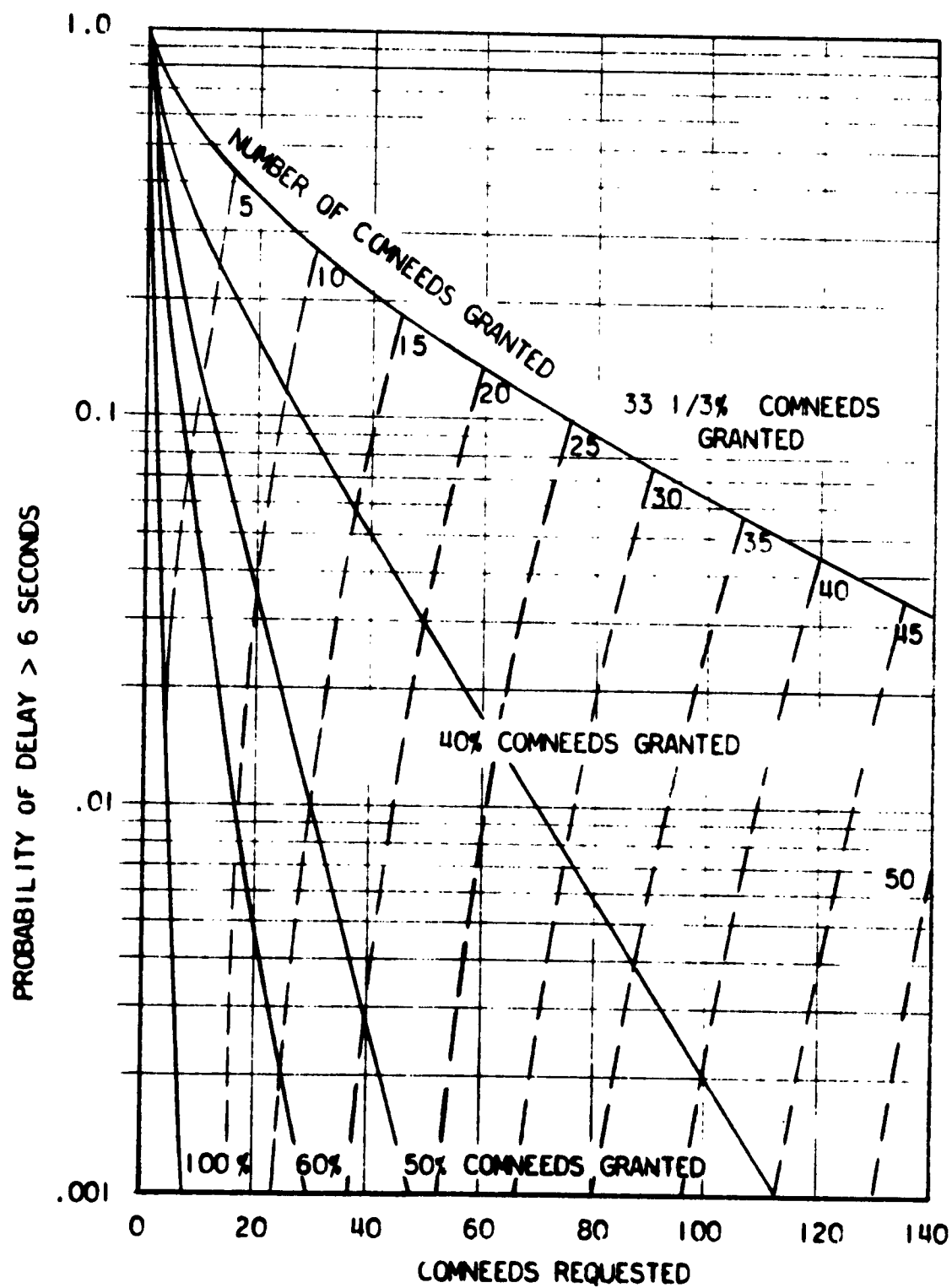


Figure 2. Probability of Delay > 6 Seconds as a Function of Comneeds Granted Initial Trunk Loading $a/c = 0.25$

since they are theoretical; hence, at least two levels of precedence should be provided. Based on the standard military definition of precedence the assignment within the system would be as follows:

Z	Flash	}	A
Y	Emergency		
O	Operational	}	B
P	Priority		
R	Routine		
M	Deferred		

The precedence of each message may be determined by the subscriber; the particular input device may determine the precedence automatically, or a combination of the two methods may be used with different subscribers. Regardless of how the precedence is established, the use of the emergency and higher category must be held to an absolute minimum or it loses its effectiveness.

The terminal equipment design, as discussed in this report, is based on two precedence levels. However, there is no basic limitation in the equipment proposed which would prevent expansion to several levels of precedence should it be desirable. Effects on the design caused by this are discussed in section 3. Two possible cases where this might be desirable are for use in a jamming environment, and to anticipate increases in traffic demand beyond what the link has been designed to meet. The conditions and the alternate approaches to precedence level to satisfy them are discussed in later sections.

2.6 Traffic Statistics

At this point some remarks concerning the results of the analysis of actual traffic flow, which was conducted as part of this study, are in order since they have applications in the following discussions.

An analysis of traffic data for existing AACS and SAC teletype nets was made to validate earlier assumptions and to provide information for engineering estimates in areas where message statistics are pertinent.

- (a) The average message length is 3 minutes.
- (b) The average message length in bits is as follows:

Teletype	7000
Data	20,000
Graphics	4,000,000

- (c) The precedence distribution by per cent of total volume is 50 per cent operational immediate or higher, with emergency and flash contributing 1 per cent on the average.
- (d) Time distribution density of traffic flow is unknown. Inconclusive data indicates that peaks occur at ends of calendar periods (i.e., day, week, month, year).

The general conclusion is that the analysis made in determining the reference system requirements is valid because it was based on the best information available.

2.7 Alternate Solutions (Growth Potential)

The trunk requirements, precedence levels, and definitions of reliability discussed thus far have been based on determining a maximum efficiency system and establishing the terminal equipment requirements to support it. It is clear that alternate solutions may be desirable and in fact are possible. Since the trunk requirements are based on all messages meeting the A reliability criteria, any relaxation of this to include B and C reliability can only result in reduction of trunks required. However, this is not recommended. Past experience with new systems which provide an

increase in quality indicates that they generate an increase in requirements and traffic volume which is unpredictable. Hence a safety factor in the initial design is desirable. However, to anticipate this increase it may be desirable to build in necessary system discipline to handle the excess traffic. In this instance, a redefinition of precedence levels handled is required. The following breakdown is recommended:

X } Y }	A
O	B
P } R } M }	C

where A, B, and C carry the delay criteria as previously defined. Since the building block approach has been used in the design, this alternate should present no problem.

2.8 Jamming Environment

Analysis of the jamming environment is beyond the scope of this study. However, where anti-jamming advantage is gained by the reduction of information bandwidth to transmission bandwidth ratio, it is suggested that this be most easily done by dropping traffic by order of precedence. As a basis for determining the advantage gained, the following information that is based on three levels of precedence is offered:

Precedence	% of Volume left after traffic reduction	
	Case 1.0	Case 2.0
C out	31%	60%
B + C out	1%	10%

Case 1.0 is based on traffic statistics for a normal environment. Case 2.0 is based on an estimated increase in high priority traffic due to a hot war situation. The actual case is likely to fall between these bounds.

2.9 Operational Procedures

2.9.1 General

Disciplining the message flow through the system is controlled by precedence levels as defined previously. Methods of pre-emption are considered in the design with A and B categories having pre-emption rights. Other operational procedures of importance are system monitoring, routing procedures and alternate routing provisions.

2.9.2 System Monitoring

As a means of monitoring the message handling efficiency and to provide a basis for changing operational procedures or criteria, it is recommended that all traffic flow be logged to provide traffic statistics. Periodic analysis of the data would provide a sound basis for engineering decisions to upgrade or modify the system.

2.9.3 Routing Procedures

Voice- Voice communications must, of course, be handled on a real-time basis. Their routing within the reference network will be handled by automatic switching equipment of the reed-relay type. The numbering plan will use 10 decimal digits: 3 for the area code, 3 for the central office code, and 4 for the called station number. Access to the network will be through a local PABX by pushbutton tone signalling or operator assistance in the case of conference calls. The subscriber will probably be unaware of the fact that his call is being routed through a satellite link, except in cases where satellites are unable to provide continuous coverage

between two stations. The outage times in such instances might range from several seconds to two or three minutes. If a subscriber attempts to place a call on a satellite link in which an outage exists, he will be returned a busy signal and will probably not even suspect the real reason. For calls already in progress at the time of the outage, a special audible signal could be furnished to the subscribers just prior to

the anticipated outage to warn them of the necessary interruption. It should be noted, however, that discontinuities from this cause will not occur at all in links between primary stations and will usually occur only in long-distance links involving a far-northerly station. Interruptions caused by transfer from one satellite to another will be of such short duration as to probably go unnoticed.

A detailed description of a voice call is given in section 3.4.3 of this report.

Graphics - Graphics traffic in the reference network will also be handled on a real-time basis because of the excessively large storage units required for store-and-forward operation. The desired connection will be established in much the same manner as a voice call. Pushbuttons for tone signalling will be provided at the graphics sending device. When the circuit has been completed, a signal lamp will indicate that transmission may begin. It will continue to indicate the continuity of the circuit.

Most graphics transmissions will be of several minutes duration. If an appreciable outage due to lack of satellite coverage should occur during the time of transmission, it will be necessary to retransmit the message. When a satellite transfer occurs during a graphics transmission, the need for retransmission will depend upon the significance of the resulting error. If errors or retransmission cannot be tolerated, a burst transmission scheme similar to that for store-and-forward traffic could be used. However, since the graphics information is generated in real-time, it must be transmitted at a higher rate of speed than the one at which it was produced in order to compensate for the transfer interval. The additional cost and complexity of such a method is not considered to be warranted in most cases.

Teletype - The routing of teletype traffic will be similar to that of existing networks. The passive spherical satellite reference network appears to impose no special restrictions on it. It would be desirable to keep the message format as short as possible to conserve valuable channel capacity. Since little general reduction can be made in the message text or ending, the heading becomes prime target for shortening. The heading defined in the Air Force supplement ACP 127(B)-1 (USAF Data Communications Procedures and Routing Doctrine for the Combat Logistics Network) is considerably more concise than the older form of ACP 127(B) (Communication Instructions; Tape Relay Procedures). It contains the necessary items of information for use in the reference network. Therefore, no particular benefit would accrue from the establishment of a new format especially for the reference network since the existing one has already been proven and personnel trained in its use. The continued use of this format simplifies the possible future integration of other networks into the reference network or vice-versa. ACP 127(B)-1 has the additional advantage that it can be used with other types of messages such as data and magnetic tape.

Three types of routing are specified in ACP 127(B)-1; namely, Specific Routing, Predetermined Routing, and Routing Line Segregation. In Specific Routing, the originating operator specifically indicates all relay stations through which the message will be routed. This substantially increases the amount of routing information which must be transmitted. With Predetermined Routing, the originating operator indicates only the final destination and relay stations rely upon prearranged instructions. Routing Line Segregation is a special case of Predetermined Routing used when a message is routed to several destinations. Each relay station deletes all routing indicators from the routing element except those applicable to a forward transmission. In this manner, message length is cut to a minimum and valuable transmission time saved. Routing Line Segregation is felt to be the best method for the reference network; it is also in keeping with the system prescribed for the COMLOGNET.

The following is a typical routing of a sample teletype message prepared according to ACP 127(B)-1 format by HQ, EASTAF, MATS, McGuire AFB (5,000A) for delivery to MATS Commands at Goose (55,000), Keflavik (55,400), Torrejon (70,000), and South Ruislip (7,500A). The station routing symbols used are 5 alpha-numeric ones corresponding to figure 1. Various other routing symbols could be devised. However, any system selected should be in keeping with the tree-type configuration of the reference network so that a station's location, and hence the routing of traffic to it, can be easily determined by its routing symbol.

At McGuire, the message is logged, encrypted, and transmitted to Andrews by conventional teletype land lines. The original message has the following form:

R	H	U		C	I	5	Ø	Ø	Ø	A	Ø	Ø	Ø	1		Ø	9	Ø	1	1	4		-	5	
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26

5	4	Ø	Ø		5	5	Ø	Ø	Ø		7	5	Ø	Ø	A		7	Ø	Ø	Ø	Ø		.	
27	28	29	30	31	32	33	34	35	36	37	38	39	40	41	42	43	44	45	46	47	48	49		

TEXT 2 CR's 8 LF's 4 N's

The items in the message heading are defined below. The numbers under each item for reference purposes only and do not appear in the actual message.

1. Precedence (In this case, Routine; could also be Z, Y, O, P, or M)
2. Message type designator shows input device which prepared message and format of text. (In this case, teletype.)
3. Classification (Unclassified)
4. Separator
- 5, 6. Content indicator for management and analysis purposes.
7. Separator
- 8, 9, 10, 11, 12. Originator Routing Indicator

- 13, 14, 15, 16. Station serial number to identify message.
17. Separator
- 18, 19, 20. Calendar day numbered in consecutive order from Jan. 1st.
- 21, 22, 23. Time of filing expressed in GMT to nearest tenth of an hour.
- 24, 25. Start of routing signal.
- 26 thru 48. Routing Indicators of addressees; a maximum of eight routing indicators separated by spaces may be used.
49. End of routing signal. All data following this will be text. This signal appears after last addressee; not necessarily in position 49.

At Andrews, the message is decrypted and the teletype code is converted to Fielddata code (see figure 16). The bit rate is increased, and the message is processed and switched to two outgoing links, each with a different heading in accordance with Routing Line Segregation. Following encryption and multiplexing on each link, transmission proceeds in accordance with precedence and channel capacity.

Transmission to Goose Bay

R H U C I 5 0 0 0 A 0 0 0 1 0 9 0 1 1 4 - 5
 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26

5 4 0 0 5 5 0 0 0 .
 27 28 29 30 31 32 33 34 35 36 37

TEXT 2 CR's 8 LF's 4 N's

Transmission to Lajes

R H U C I 5 0 0 0 A 0 0 0 1 0 9 0 1 1 4 - 7
 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26

5 0 0 A 7 0 0 0 0 .
 27 28 29 30 31 32 33 34 35 36 37

TEXT 2 CR's 8 LF's 4 N's

At Goose, the message is decrypted, demultiplexed, processed, and switched. For Keflavik, it is again multiplexed and encrypted for relay via satellite link. The local drop is slowed down, reconverted to teletype code, and typed out on hard copy for the local MATS commander.

Transmission to Keflavik

R H U C I 5 Ø Ø Ø A Ø Ø Ø 1 Ø 9 Ø 1 1 4 - 5
 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26

5 4 Ø Ø .
 27 28 29 30 31

TEXT 2 CR's 8 LF's 4 N's

Local to Goose

R H U C I 5 Ø Ø Ø A Ø Ø Ø 1 Ø 9 Ø 1 1 4 - 5
 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26

5 Ø Ø Ø .
 27 28 29 30 31

TEXT 2 CR's 3 LF's 4 N's

At Keflavik, the message is decrypted, demultiplexed, processed, switched, slowed down, reconverted to teletype code, and typed out on hard copy for the local MATS commander.

At Lajes, the message shown in "Transmission to Lajes" (previously mentioned) is decrypted, demultiplexed, processed, switched, multiplexed and encrypted on a satellite link to Torrejon when precedence and channel capacity permit.

At Torrejon, the message is decrypted, demultiplexed, processed, and switched. For Croughton it is multiplexed and encrypted, and transmitted by satellite link when precedence and channel capacity permit. For the local output, it is slowed down, reconverted to teletype code, and typed out on hard copy.

Transmission to Croughton

R H U C I 5 Ø Ø Ø A Ø Ø Ø 1 Ø 9 Ø 1 1 4 - 7
 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26

5 Ø Ø A .
 27 28 29 30 31

TEXT 2 CR's 8 LF's 4 N's

Local Output at Torrejon

R H U C I 5 Ø Ø Ø A Ø Ø Ø 1 Ø 9 Ø 1 1 4 - 7
 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26

Ø Ø Ø Ø .
 27 28 29 30 31

TEXT 2 CR's 8 LF's 4 N's

At Croughton, the message is decrypted, demultiplexed, processed, switched, slowed down, reconverted to teletype code, encrypted, and relayed via conventional teletype land line to South Ruislip.

At South Ruislip, the message is decrypted, logged, and typed out on hard copy for the local MATS commander.

Data - Routing procedures for data cards are described in ACP 127(B)-1. These are similar to the teletype pattern and are suitable for use in the reference network.

ACP 127(B)-1 specifies that only unclassified messages are to be transmitted. This specification is due to the fact that the data transceiver circuits utilized in the COMLOGNET are not secured circuits. However, the use of ACP 127(B)-1 procedures will not affect the security of the reference network transmission if proper encrypting and decrypting facilities are included in it.

The occasional lack of a satellite transmission path on some links will require the use of storage devices to hold outgoing data and teletype messages until transmission can be resumed. In addition, it is proposed in section 3.2.2 of this report to interrupt these types of messages at regular intervals to permit transfer from one satellite to another. The fact that these messages may have to be automatically delayed for a time due to either reason will not affect their basic routing procedure or format.

2.9.4 Alternate Routing

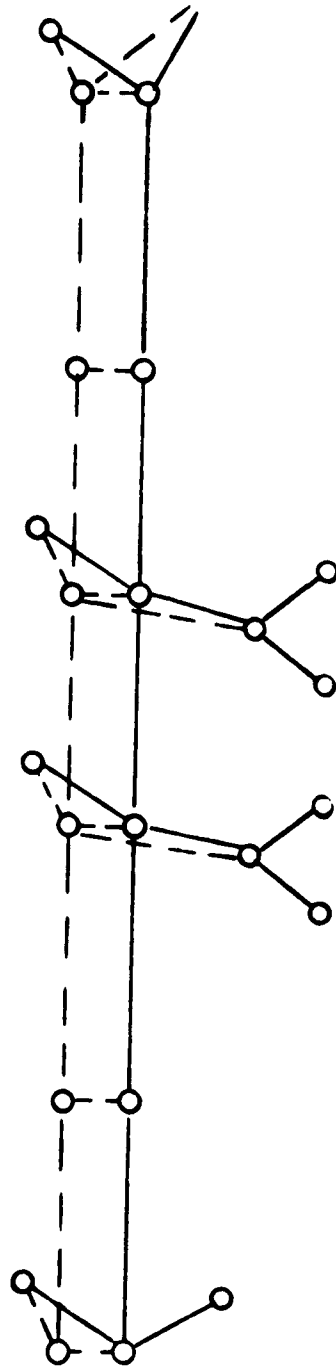
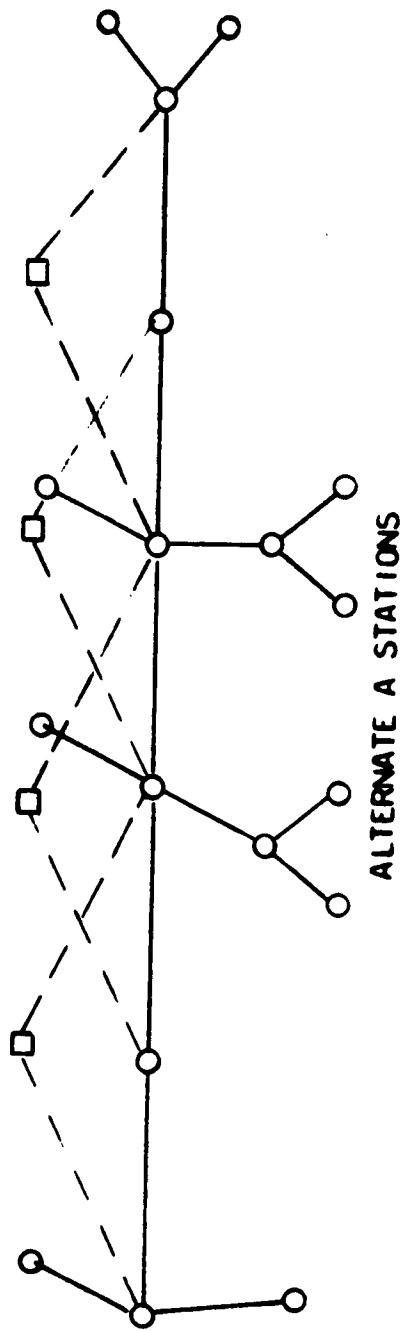
The concept of alternate routing in a passive spherical satellite network is somewhat different from that of a conventional system. With land lines, alternate routing is required primarily when one of the trunks between stations develops a fault, such as an open or a short, or when all the most direct trunks are busy. In such cases, a grid type configuration is usually used which provides many possibilities for alternate routing. Alternate routing in the present system is the rerouting of messages by a route within the system other than the primary route prescribed in the predetermined routing plan. The conditions which cause a need for rerouting are loss of primary stations or loss of successive satellites in the orbit. The former loss can cause catastrophic failure of the system while the latter results in degradation of service.

In the reference network, which has been developed as a tree, three types of alternate route provisions may be made.

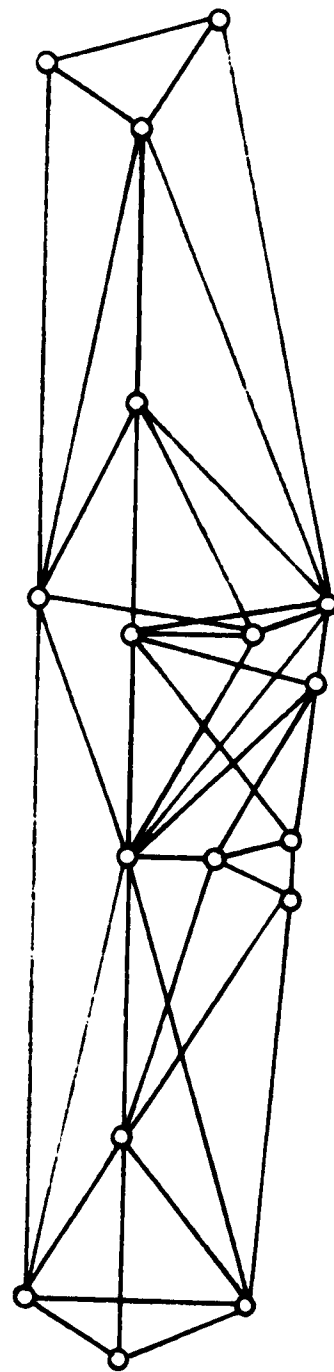
- (1) Assigning alternate A stations
- (2) Establishing a parallel tree
- (3) Developing the primary system as a grid.

These types of networks are shown schematically in figure 3. The grid system approach is the most efficient since it provides for multialternate routing. However, early in the program this was ruled out on the basis of economics. In this type of approach each station in the net would have to assume the essential configuration of the proposed A stations. The second approach of providing an alternate primary route is a compromise of the grid system approach where the A stations are interlinked and linked to the B stations. The obvious advantage of this is maintenance of the primary trunk through redundancy under conditions of loss of successive stations; this technique would require the duplication of A stations. Two approaches to the physical location of these stations can be considered: paralleling the alternates with the primary A stations with separations on the order of 100-200 miles; inserting the alternates between successive primary stations. The second approach provides more flexibility but may be physically difficult to achieve in some sections of the network. The third approach is the assignment of alternate A station capability to B stations already in the net. This is a compromise of the alternate primary trunk approach in that the station locations are removed from the nominal latitude of the primary trunk.

The technique of assigning B stations as alternates is recommended as the initial approach despite the attractiveness of the other approaches discussed. The locations of alternate stations are shown in figure 4. This recommendation is based primarily on economic factors with consideration of growth potential. Since the B station is required for other reasons, designating it as an A station alternate requires only the expansion of equipment rather than the addition of complete stations.

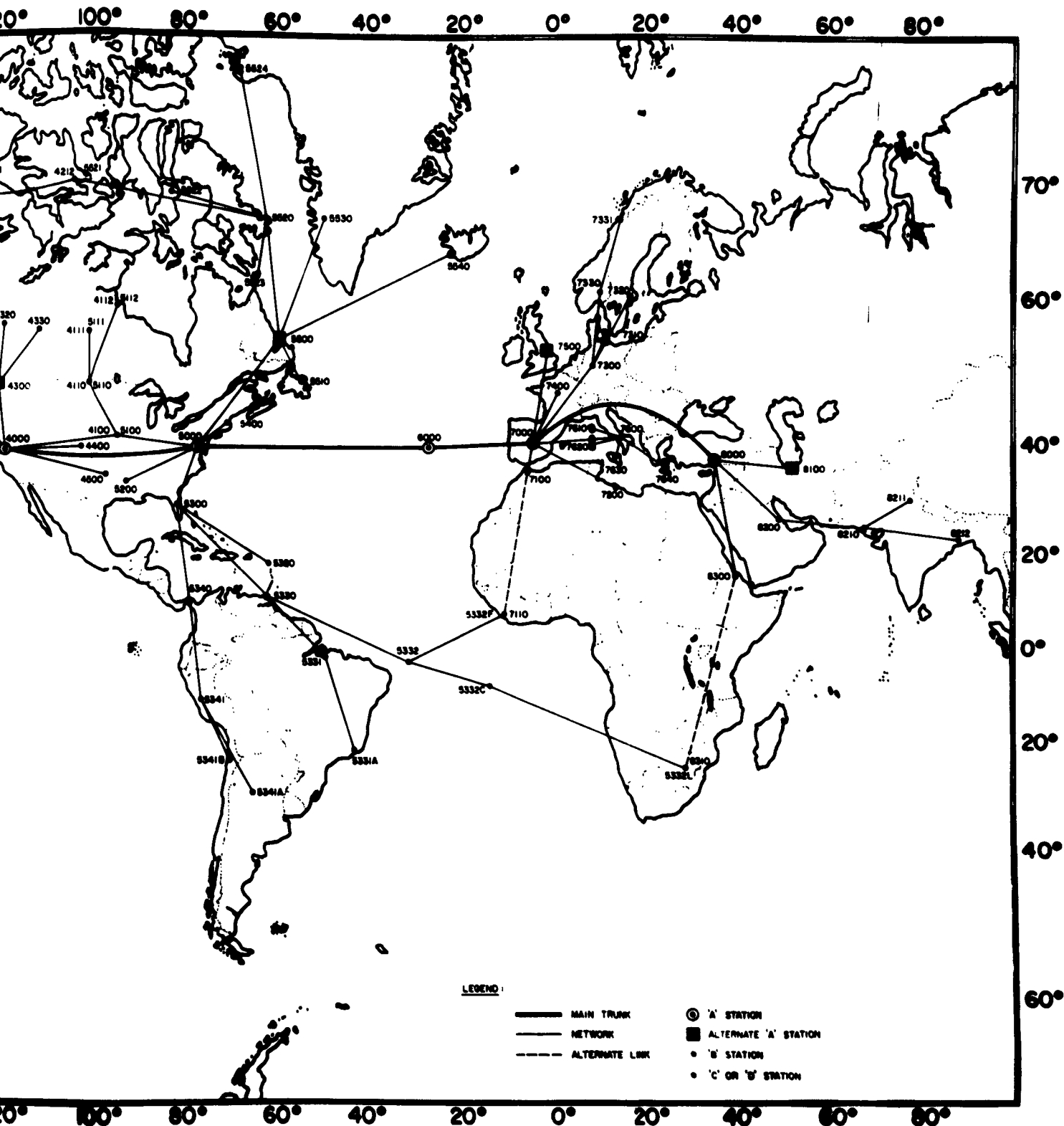


ALTERNATE PRIMARY TRUNK



GRID SYSTEM

Figure 3. Alternate Route Networks



2

Figure 4. Alternate Routing Plan

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While the topological locations are not ideal, an approximation of the alternate primary trunk is made with its inherent advantages. Hence the approach to the more desirable operational concept is available without significant increase in cost of the initial system.

Some consideration of the implied effects on the B station implementation as a result of being designated alternate A stations is in order. Two approaches are possible. It may be desirable, in the interests of economy, to equip the B station on the basis of its normal traffic handling requirements. In this case the excess traffic occurring when the B station acts as an A station is likely to cause an overload. Since precedence and pre-emption capability has been built into the system, high precedence messages would normally be expected to be handled with acceptable delay. Lower precedence messages would be handled with longer delays than normal. Under this condition, the least that would be required would be an increase in storage capacity, above that normally required, to anticipate this delayed traffic.

The second approach would be to equip the alternate A stations with the full additional capacity to handle the normal A station traffic. This in effect, would result in an alternate primary trunk for the system.

For the initial approach the recommended solution is a compromise in which the alternate A station is provided with sufficient equipment to assure that all necessary trunks can be established, all high precedence traffic can be handled with acceptable delay, and sufficient storage is available for the delayed traffic.

3. TERMINAL EQUIPMENT REQUIREMENTS

3.1 Philosophy of Design

Before discussing the specific requirements for the terminal equipment in the reference network, a method of operation shall be outlined. In any communications network, it is the function of the terminal equipment to provide a connection between two subscribers, or users, of the network. In a very simple system, this may be nothing more than a manually-operated patchboard. In a complex, long-haul communications system the functions of the terminal equipment may include switching, routing, sorting, storage, conversion, and multiplexing.

The network, as envisioned here, will handle voice, graphics, teletype, and data traffic for a selected group of subscribers located throughout the world. This combination of traffic includes signals in both analog and digital form and presents a major problem to the design of the terminal equipment. A choice must be made between a hybrid system, which will handle each type of traffic in its original form, or an integrated system, with all incoming traffic converted to a unified form. The hybrid system would accept incoming signals from a demultiplexer, with each signal form requiring its own type of line equipment and common equipment. If this global network were being implemented today, it would likely appear in this form.

For a network proposed for the 1965-1970 period, however, an integrated system appears to be a realizable goal. The conversion of all signals into digital form allows a building-block approach to the terminal equipment design, where all stations in the network can be constructed from the same basic modules. High-speed transmission of digital signals reduces the amount of line equipment required, since many messages will time-share one set of equipment. Additional advantages of digital transmission are error-checking and the encryption of voice signals. For these reasons, therefore, digital transmission has been selected for use in the reference network.

Although the reference network is considered to be a communication system still several years in the future, it may be desirable to use or tie it into existing communication facilities, particularly during its inception. For this reason, the terminal equipment has been made compatible with present-day speeds of teletype, data, and graphics transmission. If it later becomes desirable or necessary to use faster speeds, the system can be modified accordingly.

The previous points are concerned with the traffic to be carried by the proposed network. The remaining area of interest is the effect of the transmission system upon the terminal equipment. In a low-altitude, orbital system, the transmission must be transferred from one satellite to another approximately every twenty minutes. If both satellites involved in the transfer are being tracked simultaneously, then the transfer is accomplished by switching communication channels from one antenna system to another. If transmission is continuous and transfer occurs when the two transmission paths are not equal, then errors will occur in the bit stream due to differences in time delay. Three solutions to this problem have been considered. The obvious solution is to anticipate transfer and cease transmission during the interval. This, however, is inefficient in that it causes delay in traffic handling and increases equipment requirements. A second solution is to transmit the information in bursts rather than continuously and to perform the transfer during an interval of non-transmission. This eliminates delays and compensates for delay differences. The ideal solution is to accomplish the transfer at a time of equal path length, thus eliminating the problem as far as the terminal equipment is concerned. The latter two solutions -- burst transmission

and transfer at a time of equal path length -- have been considered in the design of the terminal equipment.

3.2 Satellite Factors Influencing Terminal Equipment

3.2.1 Orbits and Coverage

The satellite orbital configuration was determined by the General Electric Company and is described in detail in RADC-TN-60-287.¹⁵ Briefly, it consists of three circular orbits spaced 120 degrees apart in longitude (measured at the right ascension) with eight satellites per orbit. The satellites are equally spaced within each orbit, but not necessarily synchronized between orbits. Orbit inclination is 43 degrees and satellite height is 2,200 statute miles.

The orbital configuration was selected to give optimum coverage to the main trunk stations. In fact, the minimum overlap time between any two "A" stations is 3.3 minutes. This overlap may be expected to decrease over a period of months due to perturbation effects. When it reaches the point where it is no longer possible to acquire the rising satellite before the setting satellite disappears, additional satellites will have to be launched if continuous transmission facilities are to be provided.

Initially, continuous satellite coverage can also be expected on all A-B station links; however, extreme northerly links will have outages due to the lack of a suitably located satellite.

3.2.2 Satellite Tracking and Transfer

Two approaches have been considered for establishing and maintaining the various satellite links necessary for the reference network. A first approach is for each station to have a transmitter, receiver,

transmitting antenna, and receiving antenna for each link in which it participates. This uses a minimum bandwidth per antenna but obviously requires numerous carrier frequencies and a large amount of equipment at any station which is included in several links. Each transmitter-receiver antenna group tracks the satellite suitable for its associated link.

A second approach is based on a general characteristic of the orbit configuration. In general, there are never more than two satellites available to a station at any one time, suitable for establishing its links. Hence, transmissions destined for several stations, each with a different geographic location, may use the same satellite as a medium. Based on this, the minimum number of antenna systems required can be equal to the number of satellites available and traffic for several links combined for transmission. An additional antenna also would be required for locating a "rising" satellite for transmission. With this approach the station requires a minimum of one receiver per link, three transmitters and antenna systems (two plus the spare for satellite transfer). However, since the information capacity for a given transmission has been increased (by the sum of the combined links) an increase in gain achieved either by increased power or antenna size is required to maintain the performance. In the reference system power and antenna size limitations preclude the implementation of this approach. This approach is not considered in the present study.

In the absence of what may be termed the "broadcast" mode described above a decision must be made regarding the most efficient satellite to be used at any moment. This decision is based to a large

extent upon minimizing the number of satellites in orbit. The moment of transfer, however, can result in a time shift which creates a synchronization problem in the received intelligence in the terminal equipment.

Orbital analysis of the reference system has demonstrated that the switching of receiving antennas can generally be accomplished at the time signals received from two satellites are coincident (indicating equal length transmission paths). Equal range-sum switching creates no problems for the terminal equipment by virtue of eliminating delay variations.

However, for the general case, it appears that there can occasionally be instances in which the signals received from two satellites will never coincide because the transmission paths do not become equal in length. Since the occurrence of non-equal range is a possibility, a solution to this problem is considered in this study.

Non-equal range sums are indicated in figure 5. Two stations, A and B, are shown with their common coverage zone. Satellite No. 1 is about to leave the coverage zone while Satellite No. 2 has just entered it. The transmission path via No. 1 is long and getting longer. The path via No. 2 is near its minimum and will soon start increasing. It can be seen that it is possible to have a case where the transmission paths will never be equal in length. This case is shown in figure 6. The more common case in which equal length paths occur is shown in figure 7.

The problem of equal length transmission paths may also be considered from the point of view shown in figures 8 and 9. In figure 9, we have two stations, A and B, in the satellite communications

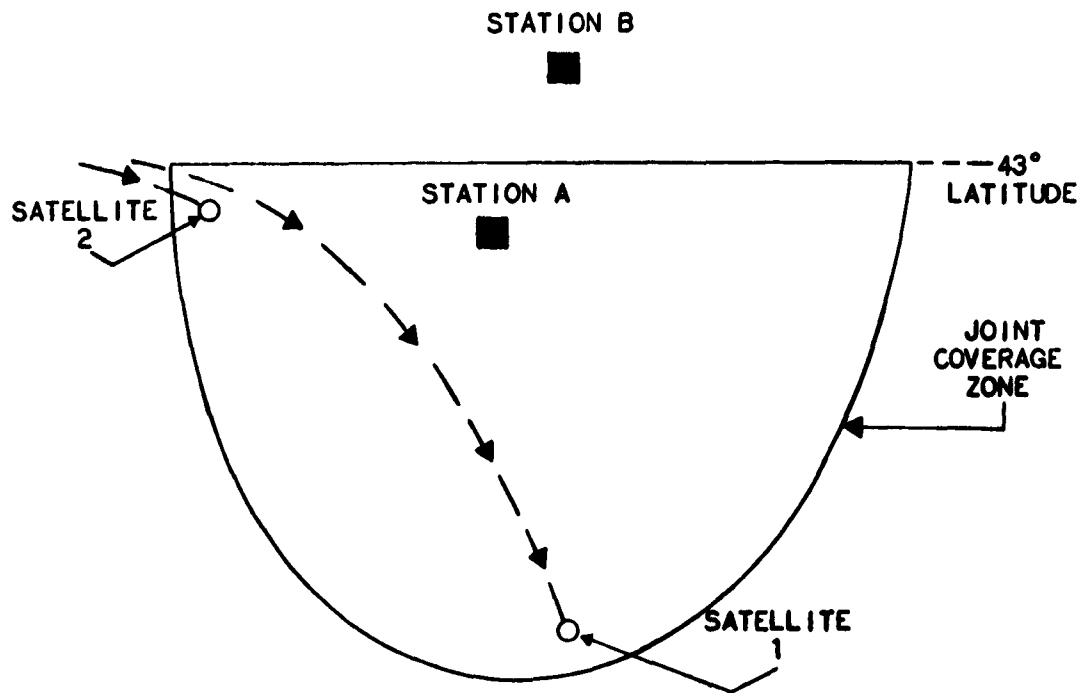


Figure 5. Satellite Course Through Joint Coverage Zone

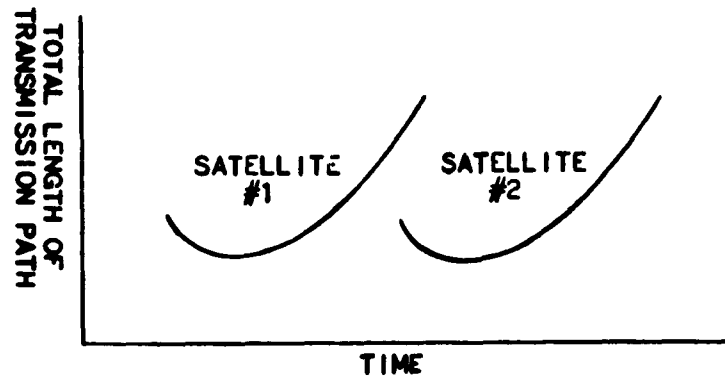


Figure 6. Transmission Paths Without Point of Equal Length

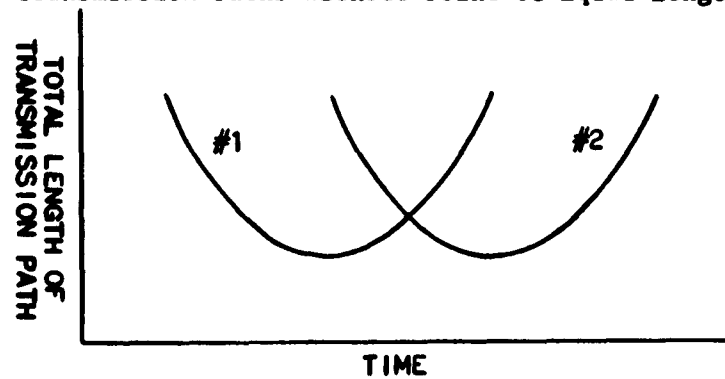


Figure 7. Transmission Paths With Point of Equal Length

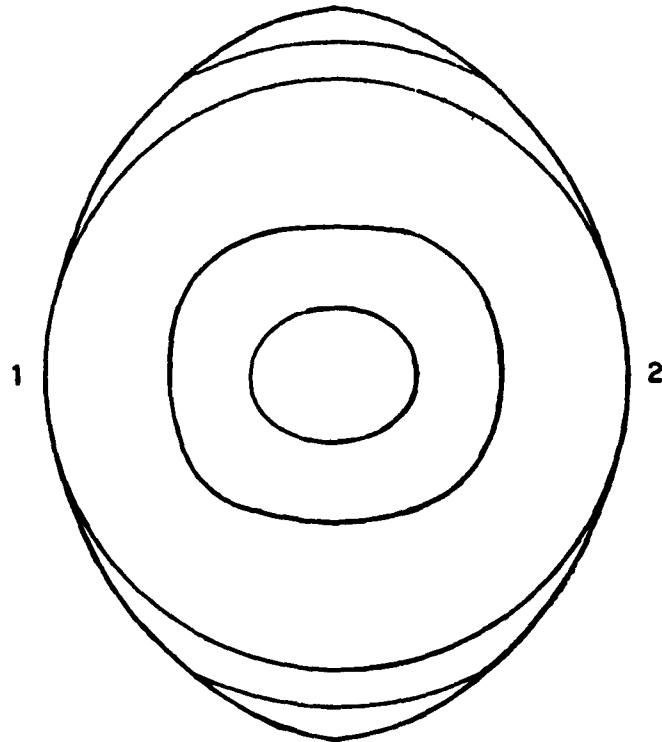
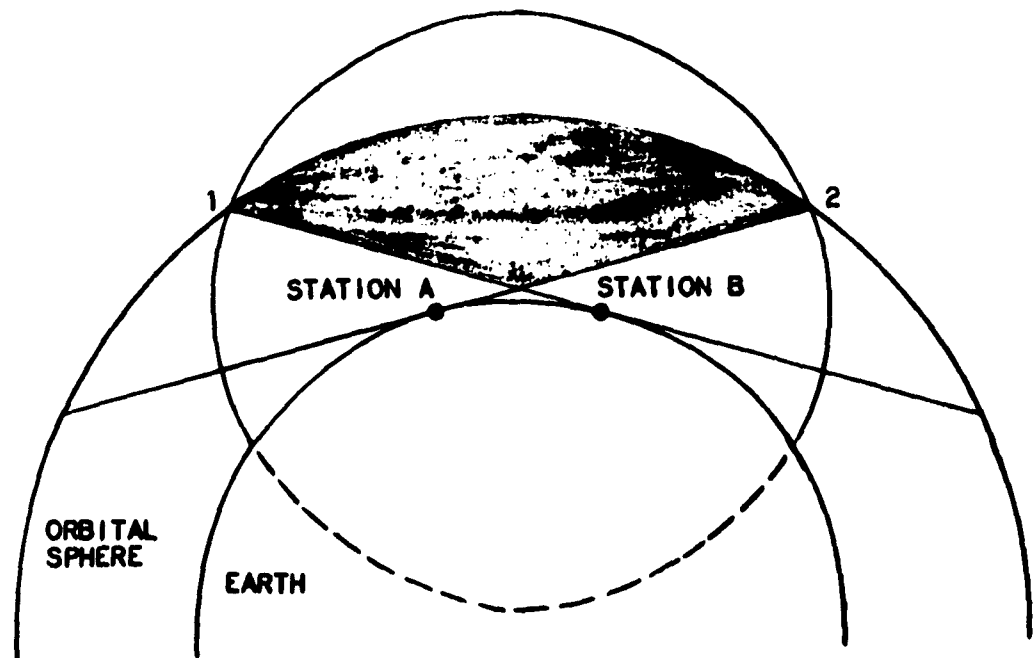


Figure 8. Joint Coverage Zone Showing Intersections With Ellipsoids



JW

Figure 9. Joint Coverage Zone for Two Stations

A 2186295

network. The orbital sphere is shown representing the locus of all satellites. For simplicity, the angle of view for each station is assumed to be from horizon to horizon, which results in a joint coverage zone consisting of a portion of the orbital sphere indicated by the shaded area in figure 9, and the football-shaped area in figure 8. This represents one constraint on the coverage area.

Another constraint is that transfer from one satellite to the next occurs when the range sums of the two paths are equal. An ellipsoid is the locus of all points, the sum of whose distances from stations A and B is a constant. As the value of the constant is changed, by small increments we produce a set of confocal ellipsoids, all having stations A and B as their foci. A small portion of these many ellipsoids will intersect orbital sphere within the joint coverage zone. The ellipsoid passing through points 1 and 2 forms a boundary condition. All ellipsoids smaller than this, but still intersecting the orbital sphere, will have a continuous line of intersection within the joint coverage zone. All ellipsoids larger than this one, but still intersecting the joint coverage zone, will have a discontinuous line of intersection within the joint coverage zone. If two satellites lie within the joint coverage zone, a time of equal path length may or may not occur, depending upon whether or not they both cross the intersection of the same ellipsoid simultaneously. To satisfy both the coverage and time delay criteria for all orbits reduces the size of the joint coverage zone to the area bounded by the intersection of the ellipsoid through 1 and 2 if equal length paths are to be required.

Calculations, for example, with two stations separated by 2000 miles and an orbit altitude of 2,200 miles, give the following results. Assuming a look angle of 90 degrees for each station and the stations in an east-west line, the joint coverage zone is formed by the intersection of two spherical segments as shown in figures 8 and 9. In this example, the joint coverage zone extends approximately 90 degrees in the north-south direction and 71 degrees in the east-west direction. With a requirement for equal path lengths, the extent of the coverage zone in the north-south direction is approximately 68 degrees of arc on the orbital sphere.

Within the reference system it has been shown that transfer from one satellite to the next will normally occur at the instant of equal length transmission paths. However, since it may be possible with some orbital configurations to occasionally experience a situation where the transmission paths never become equal, an alternate method is now proposed which is not based on receiver transfer at the instant of signal coincidence. Instantaneous transfers without coincidence would result in a break or repetition in the received signal depending upon whether the new path was shorter or longer than the old. This would probably go unnoticed in voice communications but would cause errors in data, graphics, and teletype, or in encrypted information. A computer could be used at each station to determine when satellite transfer would be necessary on each link and accomplish this by interrupting transmission. As a simpler solution to this problem, it is recommended that store-and-forward information be transmitted in blocks, independent of the message length. In the intervals between block, voice, graphics, and

control channel (see section 3.3.3) transmission could continue. Voice, data, and graphics could be handled in a similar manner. For real-time transmission an alternate speed-up and wait, and slow-down and fill-in procedure would be necessary. The net delay in transmission would be approximately 25 ms per tandem section of the system traversed. All satellite transferring of receivers would occur during these intervals. The transmission of framing pulses in the control channel during the transfer interval would enable the receiving station to synchronize with the signal received from the rising satellite before the arrival of the next block of store-and-forward information.

Transfer must by definition occur when two satellites are simultaneously visible from a single station. The duration of time during which this is possible ranges from more than three minutes for newly launched satellites over primary trunk stations to zero or less after perturbation effects or at far northern stations. It is proposed that a transfer interval be inserted in the store-and-forward channels at approximately one minute intervals. This represents a compromise between more frequent transfer intervals which would be useful for shorter satellite overlaps and less frequent transfer intervals which would provide more efficient time utilization of the transmission path. The minimum overlap possible with this scheme is equal to the sum of the transfer interval spacing and the transfer interval length.

A transfer interval of about 25 or 30 milliseconds is recommended so that the delay due to the maximum possible difference in transmission paths can be accounted for and so that synchronization can be re-established before the resumption of store-and-forward traffic

transmission. The longest possible transmission path occurs when the satellite is at the minimum elevation angle of 5 degrees above the horizon as seen from both stations. The theoretical minimum length transmission path is with two coincident stations viewing a satellite directly overhead. The distances, as determined mathematically, in Appendix V, for these two cases are 8,800 and 4,400 miles respectively. For such a satellite transfer, this represents a maximum change in transmission path length of 4,400 miles which is equivalent to a propagation time of 23.7 milliseconds. The additional time required to re-establish synchronization will determine the necessary length of the transfer interval.

A special character can be transmitted in the control channel (see section 3.3.3) to indicate the start of the transfer interval. The receiving station will continually examine the received signal for the presence of this character. When detected, the transfer indicator will be stored by the station for a period of approximately 50 milliseconds. During this time if no similar signal is received via a second satellite path, this particular transfer indicator will be discarded. The same procedure will be repeated each minute. When a transfer indicator is received by a station from two satellites within the same 50-millisecond period, the station will immediately transfer from one satellite to the other on that link, upon receiving the second signal. The second satellite need be monitored only when a transfer is imminent. This can be controlled by the orientation of the receiving antennas. Acquisition of a rising satellite will be by means of orbital elements distributed periodically to all network stations from a central source or by a

programmed search derived from locally calculated orbital elements derived from previous passes.

In links where continuous satellite coverage is not available, the situation is slightly more complicated. This will occur most frequently, if not entirely, in the lowest echelon links. Verbal discussions with the orbital system contractor on this study program indicate that the greatest outages due to lack of satellite coverage will be no more than 15 minutes three times per day. At these stations, communications will be possible only when the satellite is in certain portions of the sky. As the transmitting antenna approaches the boundaries of these areas, it is suggested antenna pointing data be analyzed and a signal sent to the switching equipment. A special tone will be inserted on all telephone circuits to warn subscribers of the impending interruption. Calls dialed at this time will receive a busy signal. When the boundary limit is reached, all voice and information transmission will cease. It will not be resumed until another satellite appears within the necessary area.

3.3 Types of Signals

3.3.1 Separation of Traffic

The reference network must be capable of handling voice, teletype, data, and graphics traffic. Ideally, it would be desirable to transmit all of these on a real-time basis. However, this would require a large number of trunks during the busy hours, many of which would stand idle during the slack periods. It would be considerably more efficient if some of the busy-hour traffic could be delayed until a slack period. In addition, a real-time message which traverses several links must have a trunk available simultaneously on each link. But, if the transmission on each link can be delayed until a trunk becomes free, a significant reduction in the number of required trunks can be achieved. Thus, it is seen that, particularly in a satellite system with several links, the use of store-and forward operation wherever possible is desirable.

By its very nature, voice traffic must be handled on a real-time duplex basis. The other three types of reference network traffic lend themselves to store-and-forward operation. However, the average military message lengths as determined by the Unicom project 35 are as follows:

<u>Message Type</u>	<u>Average length in bits</u>
Teletype	7,000
Digital Data	20,000
Graphics	4,000,000

Consequently, this means that to handle graphics on a store-and-forward basis, the storage capacities must be about 200 times greater than is necessary to handle only teletype and data. To eliminate excessively large stores, it is recommended that in the reference network, graphics be handled along with voice on a real-time basis.

Another question to be resolved is whether traffic should be transmitted in all digital form or in a combination of analog and digital. The input from teletype and data sources will be digital; conversely, inputs from voice and graphic sources will be analog. A solution to the problem may best be arrived at by examining the advantages and disadvantages of each transmission technique.

<u>Technique</u>	<u>Advantages</u>	<u>Disadvantages</u>
All Digital	Secure Encryption. Signal regeneration at relay points without a reduction in signal-to-noise ratio. Ease of time-division multiplexing and electronic switching cross-walk removal by slicing and re-covering may be easily combined into one information channel.	Wide bandwidth required for digitized voice (system can operate at lower S/N ratios).
Combination of Analog and Digital	Analog Signals would not have advantages mentioned above for all-digital technique. Information is separated into two distinct channels.	Voice channels only require 4KC bandwidth each.

From the comparison chart above it is apparent that the only disadvantage to an all digital system is the wide bandwidth required for digitized voice. Although the base bandwidth is greater, the signal-to-noise ratio required is considerably more for analog signals than digital. While a S/N ratio of 20 db may be tolerated for digital information, 60 db is required for toll quantity analog channels.

With this fact in mind and from the above comparison the advantages of an all digital system outweigh the disadvantages of a wider bandwidth requirement. Therefore it is recommended that the voice and graphics traffic be converted from analog to digital form.

3.3.2 Multiplexing

There are two basic types of multiplexing, time-division and frequency-division. In time-division multiplexing each of the "n" participating channels shares the total available bandwidth for $1/n$ of the time. In frequency-division multiplexing, each channel has $1/n$ of the available bandwidth for the entire length of time.

Both types of multiplexing can be used with either analog or digital signals; however, the nature of digital signals makes them especially suited for time-division multiplexing. In the reference network, it is recommended that both time-division and frequency-division multiplexing be employed. Utilizing the all digital transmission technique enables the four types of traffic from each link to be combined by time-division multiplex into one binary bit stream. Then, the binary streams for the various links in which the station participated could be combined by frequency-division multiplexing into a single output. An alternate proposal is to use two independent binary bit streams, one high speed stream for real-time

traffic (voice and graphics) and one lower speed stream for store-and-forward traffic (data and teletype). The single binary bit stream method is recommended for use in the reference system. Justification for this choice is given in Appendix II.

3.3.3 Composition of the Bit Stream

The single binary bit stream recommended for use in the reference system is as mentioned above, a time-division multiplexed signal of all four traffic types flowing over one link. The bit rate is specified by the following formula:

Total bit stream rate (bits/second) = 38,400 bits/second (No. of voice channels + 3)

Where

38,400 bits/second is the optimum PCM bit rate for transmission of the various traffic types found on the link. The reasons for this choice are given in Appendix III.

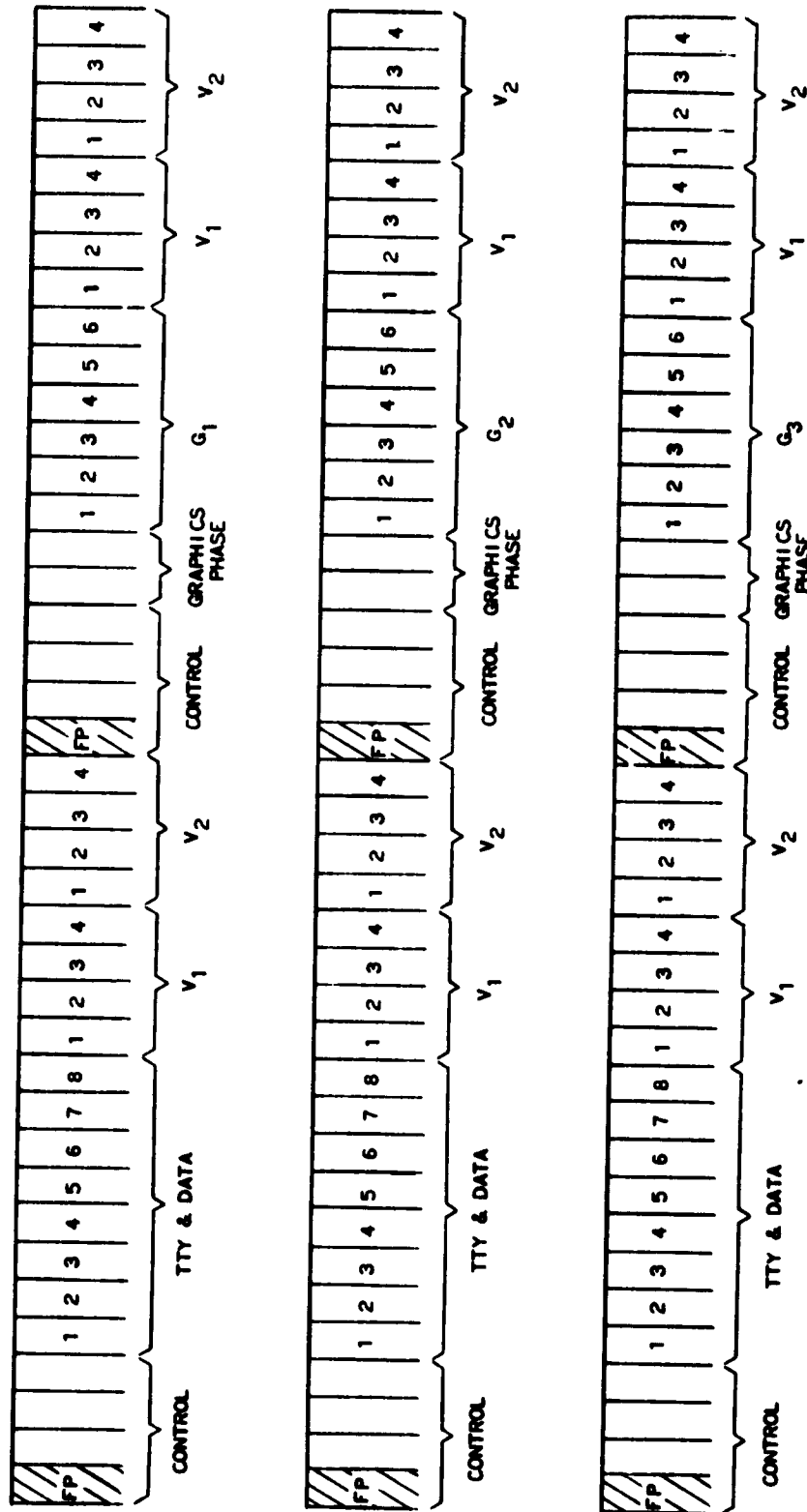
It can be seen that the total bit stream rate is dependent on traffic requirements. The rate varies from link to link but remains constant for any one link. The basic bit rate for each voice channel in a simplex link is 38,400 bits per second. Two additional channels of 38,400 bits per second are needed to carry the graphics, data, and teletype traffic.

A third channel width of 38,400 bits per second has been added in each simplex link for control purposes. This effectively increases the link bit rate by the factor

$$\frac{\text{Voice Channels} + 3}{\text{Voice Channels} + 2}$$

The control channel appears first in each frame interval, and the first bit in the channel serves as a framing and sync pulse. By alternating regularly between the "one" state and the "zero" state, the presence of this pulse can be detected by filtering and the information used to extrapolate the positions of the succeeding information bits within the frame. If it should fail to be detected during a frame interval, extrapolation of bit positions could continue from the previous frame.

One possibility is shown in figure 10 with the total bit stream rate of 192,000 bits/second. Its capacity is two voice channels, three graphics channels, and any combination of data and teletype channels which will result in 38,400 bits per second. How this may be achieved will be discussed further in section 3.5. Since the data and teletype uses an eight-bit code (twice the four-bit code of digitized voice), two voice channels are assigned to them to avoid splitting a character. This means that twice as many bits are transmitted per frame as should be. To compensate for this, only every alternate frame is used for teletype and data. The channel space in the remaining frames is then available for graphics traffic. However, because of the slow speed of graphics transmission (see section 3.6.6) only every sixth frame need be used for each graphics channel. This allows three graphics channels to be used alternately with the data and teletype. In addition, the graphics transmission uses only a six-bit code. In a two voice channel space this allows for two extra bits which are used to indicate the phase of the graphics channels. If additional graphics capacity is required, two voice channels can be added which will provide six more graphics channels. This type of detection requires a short time interval to establish



FP - FRAMING PULSE
2 VOICE, 3 GRAPHICS, AND 1 STORE-AND-FORWARD CHANNEL
192,000 BITS PER SECOND

Figure 10. Composition of Sample Bit Stream

synchronization. It would be desirable to detect the sync pulse immediately by its uniqueness rather than by its repetition rate. This might be accomplished by making it wider than the other pulses, or if frequency shift keying is employed, a separate frequency could be used for the sync pulse. This is a problem to which no wholly satisfactory solution has been obtained. Further research in this direction is indicated.

The remaining bits in the control channel perform additional functions. For instance, the store-and-forward information contains an

extra bit in each character for parity check. The receiving station checks the parity of the incoming signal in blocks of 32 bits. The result of this check must be returned to the sending station to acknowledge that the block was correctly received or to request a retransmission. Since it is assumed, in the reference network, that there will be a return bit stream for each outgoing bit stream, part of the control channel may be used for this purpose (see Appendix IV).

It is recommended that continuous transmission be maintained on all links whenever possible regardless of traffic volume. This will serve two purposes; it will enable more rapid detection of system faults during slack periods, and it will prevent an enemy from gleaning any clue to military operations by means of traffic volume analysis. Uniform traffic flow can be accomplished by the use of simulated traffic when no real traffic is available. It is necessary to notify the receiving station that simulated traffic is being transmitted so that it can reject it. This can also be done by use of bits in the control channel.

During the satellite transfer intervals, no store-and-forward information will be transmitted. To avoid the appearance of gaps in the assembled message, any of the blank blocks transmitted during this interval must be discarded. These could be identified by use of additional bits in the control channel.

3.4 Real-Time Traffic Handling

3.4.1 State of the Art

Among the many development programs currently being undertaken

in the transmission and switching of voice signals in digital form are: the Bell System ESSEX (Experimental Solid State Exchange); the Stromberg-Carlson AN/TTC-12, 13, and 15 Electronic Switching Systems; the Digicom Delta-Modulation System developed by ITT; and the Unicom System being developed for the U. S. Army. These, as well as some additional systems, are reviewed in more detail below. While none of them is directly suitable to the satellite reference network, many of the techniques employed are applicable.

Essex II - The Experimental Solid State Exchange (Essex II) demonstrates an entirely new approach to telephone switching and transmission by use of speech digitization and time division switching.

There are four important, inter-related points involved in the Essex concept. The first is that line concentrators are used near the customers' premises to reduce the number of wires connecting to the central office. Second, switching at the concentrators is time-divided. Third, pulse code modulation is used for transmission between the concentrators and the central office. Fourth, the PCM signals are connected directly through the central office by time-division switches. They are reconverted to standard voice-frequency signals at the concentrator near the called customer.

Each concentrator is capable of handling 255 voice-frequency telephone lines. From the concentrator to the exchange, 24 PCM channels are provided on a time-division basis. Twenty-three of these are for voice conversations and the 24th is reserved for detecting new requests

for service. A seven-bit digital code permits 128 speech levels. An eighth-bit is used for control signals to the exchange. Transmission rate is 1,536,000 bits per second, which necessitates regenerative repeaters every 6,000 feet for 22 gauge conductor pairs. Four-wire transmission is used - one pair for each direction. A third repeatered pair is used for synchronization and control. A trunkor is used to connect an ESSEX type exchange to the voice-frequency trunks of existing exchanges, but no mention is made of the possibility of connecting together two ESSEX type exchanges.

AN/TTC 12, 13, 14 and 15 - The AN/TTC 12, 13, 14, and 15 are Army switchboards under development for tactical field use. They are used in a four-wire subset-to-subset system. There is no dc signalling. Calling and clearing is by tone number set up by multifrequency tones. The transmission objective is a nominal 4 kc audio-frequency band with characteristics to make it suitable for medium speed data transmission up to about 2000 bits per second. Reed relay crosspoints are used on the Number 14 board which would handle anything from dc to 100 kc. The frequency limitations are essentially a matter of line transformer problems. The control principles are straightforward and do not involve any new techniques or basic principles.

The switching network of the Numbers 12, 13, and 15 boards is based on a 32 channel PAM system with a sampling rate of 12.5 kc per channel. The trunking is based on a fixed time slot per line and interconnection is by audio links having full availability access to all highways. The link channel memories are individual cycling devices with an 80 μ s cycle time. They are triggered in the correct phase for the wanted channel and then

locked to the master pulse generator. Coding for security reasons is difficult because of the use of PAM.

Digicom - The Digicom system is an all pulse communication system for tactical military use. Each switching center can have connected to it local loops and trunks. Each of the loops is capable of transmitting a 1 mc digital signal which is divided to produce a 24 channel pulse, delta modulation transmission path. The 25th bit is always present and is used for synchronization. In the loop, there is placed at least one channel Drop Facility which has the capability of allowing up to 12 handsets to be multiplexed into the 24 channel loop. The sampling rate of the speech is 40 kc, and any type of analog data which is compatible with this sampling rate may be sent.

UNICOM - The Universal Integrated Communications System for the U. S. Army is presently under development by Bell Telephone Laboratories. This system includes digital stations which originate all modes of information in synchronous, digital form, and analog stations which may originate calls in either analog or asynchronous, digital form. The automatic circuit exchange (ACE) is equipped with two switching matrices, a space-division matrix for calls from analog stations, and a time-division matrix for calls from digital stations. Both matrices have access to the automatic message exchange (AMX) for store-and-forward handling, and both have access to common transmission facilities. For digital speech transmission, differential PCM at 38,400 bits per second is used between the subscriber's subset and the ACE. Vocoders operating at 2,400 bits per second are used between ACE's.

Morris ECO - The Morris ECO is an electronic telephone central office currently undergoing tests at Morris, Illinois. It used small neon-filled gas tubes as switching crosspoints, seven of which break down in sequence when a voltage pulse is applied to provide the speech path. The speech retains its original analog form.

The program and translation data are stored in a Flying Spot Store which is an electro-optical, storage system. Temporary memory is supplied by an electrostatic, storage tube called the Barrier Grid Store.

A disadvantage of electronic crosspoints, such as used here, is their inability to handle the normal, ringing currents. This necessitates low-power signalling and consequently, a non-standard telephone subset.

TASI - Time Assigned Speech Interpolation is a line concentration scheme based on the fact that any ordinary conversation consists of periods of speech interspersed with periods of silence. The TASI system monitors all channels of the transatlantic cable and assigns speech to a temporarily silent channel. As soon as that conversation pauses, a different conversation uses the channel. If all channels are conversing, the conversations are periodically clipped to allow the additional conversation to be inserted. The system permits a doubling of the number of simultaneous conversations with only slightly increased distortion.

Number 5 Crossbar - The No. 5 Crossbar is a modern, flexible, switching system already manufactured in large quantities. Its operating time is as fast as can be expected with electro-magnetic controls. It can readily be adapted to priority and security identification as well as to a variety of types of

incoming and outgoing signalling. All switching contacts are of noble metal to reduce transmission losses and high-usage contacts are equipped with contact protection to reduce wear. Automatic Trouble Recording of common control circuit troubles makes for low maintenance effort.

3.4.2 Reference Network Real-Time Center

In the reference network, it is proposed that real-time traffic consist of voice and graphics, both in digital form. This will require a real-time circuit switching facility with a non-blocking array. Should it be a space-division switching or a time-division switching center? Space-division switching is the method commonly used in present day telephone systems. The circuit connections are made at the start of a call and remain continuous for its duration. This method has the disadvantages of relatively slow speed, limited bandwidth, and frequent maintenance and adjustment. Time division switching is a more recent development. In this method, the crosspoints are semiconductors which change between their conducting and blocking states many times a second depending upon the application of a control bias during the progress of a call. In this manner, the connection is repeatedly made and broken-down in accordance with time-division, multiplexing principles. This method has the following disadvantages: higher noise and crosstalk, limited bandwidth, greater variation in loss between any two given terminals of a matrix, addition of delay distortion which reduces maximum transmission speed, and difficulty of maintenance.

For the reference network, it is recommended that a combination of space-division switching and time-division switching principles be used.

The space-division connection will be established only when needed for a call. Dry reed relays will serve as the crosspoints. Even though the contacts remain closed for the duration of the call, traffic will flow through them only a portion of the time on a time-shared basis. A more detailed description of the switching equipment is given in section 3.6.2 of this report.

Time-division multiplexing is proposed for all incoming and outgoing links. Any incoming time slot shall have access to any open, outgoing time slot. This requires the demultiplexing of incoming traffic and buffer storage units to permit a change in time slot if necessary.

Since a station's incoming traffic is received over several nonsynchronous links, and since each of these links is continuously changing in length, each incoming bit stream must be reclocked to get it in synchronism with the station clock. Signals can then be demultiplexed, switched, and remultiplexed on another link.

The terminal equipment block diagram for the real-time traffic center at a reference network station is shown in figure 11. This basic diagram is equally applicable to an "A", "B", or "C" station. The distinction among the various stations will be in the number of input and output links and local trunks. Most "C" stations will originate and terminate, but not relay, traffic. Larger centers may be easily formed from smaller ones by the building-block concept.

3.4.3 Handling of a Typical Call

1. Better understand the functions of each of the blocks shown

in figure 11, a typical call will be followed on its way through the real-time center of a reference network station.

It is intended to provide each voice and graphics subscriber with a four-wire subset equipped for pushbutton signalling. This method of in-band signalling, as well as in-band supervision, has the desirable feature of bandwidth conservation. There are a number of such systems presently in use. Used predominantly in Europe is the 2 VF (voice frequency) signalling system while the U. S. favors MVF (multi-voice-frequency) of the two-out-of-six form. The latter system involves the combination of two tones for each digit to overcome the problem of false signalling due to voice currents. In a PCM system signalling could be performed by straightforward coding, where certain code combinations would be reserved for the signalling functions. This would require the development of some equipment. The supervisory function could be provided by a 2,600 cycle tone (converted to digital form) on idle lines to furnish an on-hook indication. In-band monitoring of this type would not be effective during speech. To solve this problem, the least significant bit could always be transmitted as a "one". Doing this would not affect the quality of the voice signal but would provide a monitoring signal during conversation. A transistorized tone ringer would be included in each subset to amplify "ringing" tones sent from the central office.

If it should prove impractical to furnish each subscriber with this new type equipment, conventional equipment and signalling methods could be used up to the point of entry into the reference network. From there on, conversion to digitized tone signalling, supervision, and

messages would be required.

To initiate a call, the voice subscriber lifts his handset and the graphics subscriber pushes a SEND key to indicate an off-hook condition on the send line. The off-hook condition on the receive line is indicated by a tone or by a light under the SEND key. After the pressing of the START key, the called number can be transmitted by depressing the appropriate keys in sequence. The precedence may be inserted automatically by the subset or may be selected with the pushbuttons by the subscriber. After the last item of routing information is keyed, the END HEADING key is pressed. If a numbering plan of uniform length is established, the START and END HEADING buttons may be eliminated. The selected digits are transmitted to the local PABX by means of a two-out-of-six or similar frequency code. At this point, a trunk line is selected which connects into the reference satellite communications network. Graphics traffic will use separate trunks from those of voice. All information entering these trunks is converted from analog to digital form. Graphics will use a six-bit PCM code and voice a four-bit differential PCM code.

Upon entering the reference network terminal equipment, the destination code encounters electronic equipment in the switching center which temporarily stores the information, selects an appropriate outgoing trunk, and relays the destination code to the next switching center. If no trunk is available, a busy signal is returned to the originator unless the call is of high enough precedence to pre-empt a channel already in use by a lower precedence call.

When a connection with the called subscriber has been established,

a light indicates that graphics transmission may begin. Voice communication is started by the called party in the usual manner. All transmission through the reference network is in digital form. All channels between the same two stations time-share the link. Each channel is assigned a permanent time slot, and in going from link to link, it may be necessary for a single call to change time slots. Since the bit rate in each link is determined by the number of channels in that link, it may also be necessary for a call to change speed from link to link. Both of these changes are accomplished by the buffer stores preceding the switching center. Each character is read into the store and held until the next frame interval. Frame intervals are of equal duration on all links regardless of bit rate. In the following frame interval the character is read out in the time slot and at the rate assigned to the channel in which it is being transmitted. The control pulses for this operation are switched into the buffer store at the time the circuit connection is made.

In transmission between stations, the bit streams are subjected to a Doppler effect due to the movement of the reflection satellite. In addition, bit streams arriving from different stations will not be in synchronism with each other. To compensate for both of these conditions, each received bit stream is reclocked. This process consists of continuously reading into a magnetic core storage unit in synchronism with the transmitting station clock by means of sync pulses inserted in the bit stream, and continuously reading out in synchronism with the local station clock.

After reclocking, the received bit streams can be demultiplexed into separate channels using the timing of the local clock. The data and teletype channel is directed to the store-and-forward center for processing. Each voice and graphics channel enters a buffer character store of the type previously discussed prior to being switched to the proper outgoing channel. The outgoing channel may lead to another reference network station or to a local PABX served by this station. If the call is traversing another link, a procedure similar to that described for this station is followed. If the call is terminating at this station, the digital signal is converted back to analog form before entering the local PABX.

The voice subscriber is oblivious to all the digitization, multiplexing, and reclocking which is taking place during the transmission of his call. Aside from the difference in selecting the called party, the call apparently proceeds in the usual manner. When the originating subscriber completes a voice call, an end-of-transmission signal is automatically sent to the called station and the handset is returned to its cradle. The graphics subscriber signals the completion of his call by pushing an END-OF-TRANSMISSION button. This causes all connections associated with the call to be broken down and both subscribers revert to the on-hook condition.

3.5 Store-and-Forward Traffic Handling

3.5.1 State of the Art

A message-handling facility which will meet the requirements of the reference network will be basically a data-handling system. Input

requirements vary from 75 bits per second (100 words per minute) for teletype, up to 4,800 or possibly 5,400 bits per second for high speed data. In order to handle this variety of input speeds efficiently and with minimum delay, an electronic message switching center is required. This switching center will have essentially the same functional requirements as those for any other communications network with one major exception. Because of the possibility of the temporary lack of satellite coverage and the need for satellite transfer intervals, provision must be made for the interruption of transmission without a loss of information.

Three electronic, fully-automatic, store-and-forward systems have been considered for possible use in the satellite reference network. These are all in late stages of development or early, operational stages. They are: the BIX message switching system, the Air Force (AMC) ComLogNet System, and the Electronic Data Traffic Control Complex (EDTCC) of the SAC 4652 Command and Control System.

The BIX System is limited to teletype traffic in its present form. It is conceivable that a different type of line unit, with high speed buffers, would enable a system such as BIX to handle data inputs as well as teletype. In order to handle a sufficient volume of data traffic, however, the system would require major changes in the cross office equipment. The present cross office rate of 60 kc would not be adequate to handle a large number of high speed data channels.

The following comparison illustrates the major characteristics of the switching center for the ComLogNet System and the EDTCC of the

SAC 465L System:

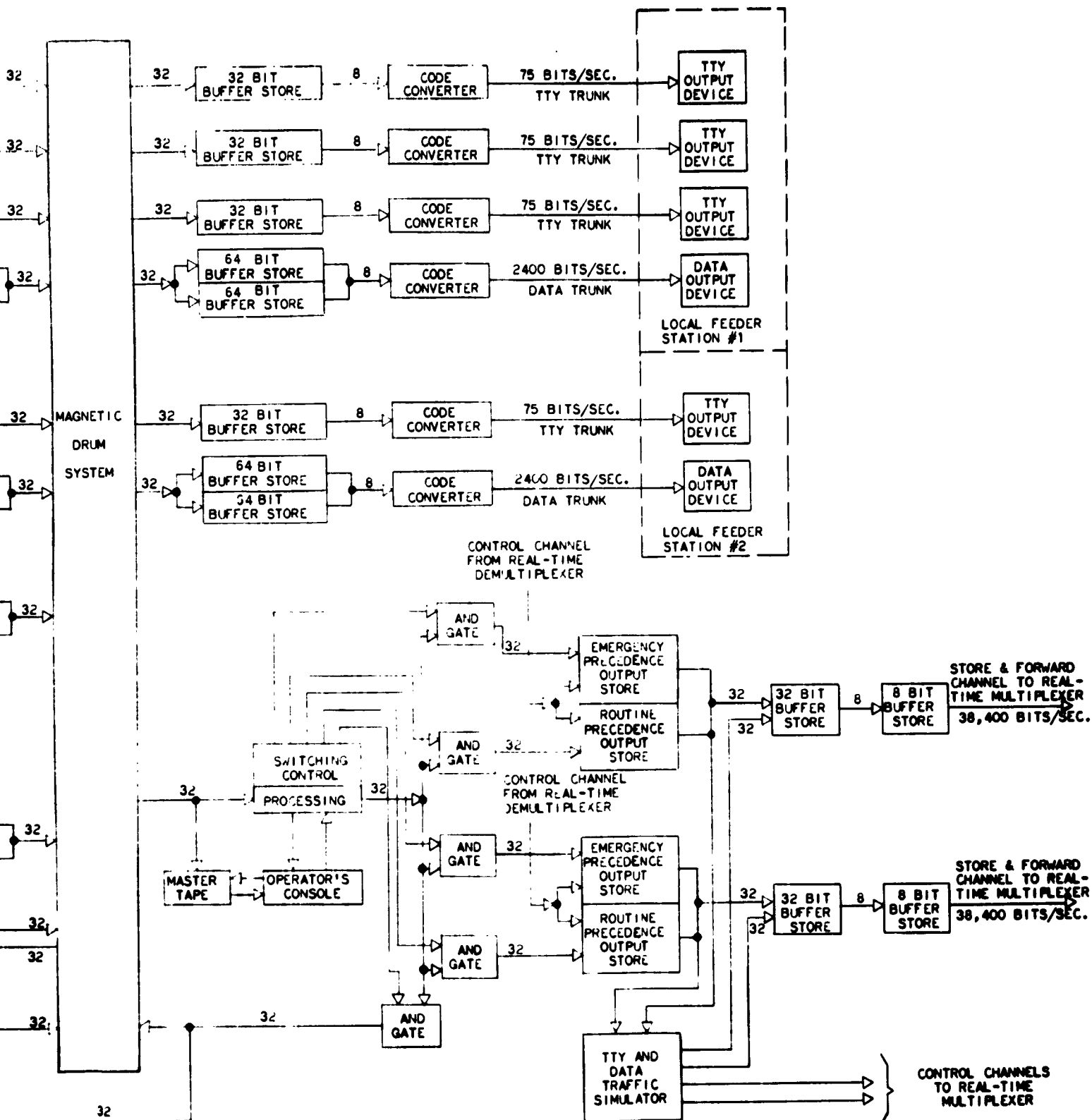
<u>ComLogNet</u>	<u>EDTCC</u>
1. 8-bit code, similar to Fieldata.	1. 8 bit Fieldata code.
2. Message records are 80 characters long.	2. Message in fixed format of 256 characters.
3. Maximum message length initially 4088 characters, eventually 40,000 characters.	3. Long messages must be broken into blocks of 256 characters.
4. Cross office rate 1 mc.	4. Cross office rate 1 mc.
5. Processing by stored program computer, Communication Data Processor (CDP).	5. Processing by stored program computer, Stored Program Element (SPE).
6. Two CDP's per station.	6. Two SPE's per station.
7. CDP employs serial operation with 10 mc circuitry.	7. SPE employs parallel operation (32-bit word) with 1 mc circuits.
8. Maximum of 100 routing indicators per message.	8. Maximum of 9 routing indicators per message.
9. Recognizes six precedence levels but classifies as High or Low.	9. Recognizes six precedence levels but classifies as High or Low.
10. Message breakthrough allowed.	10. Message breakthrough allowed.
11. Transmits special characters on idle lines.	11. Transmits special characters on idle lines.
12. Storage on magnetic drums, tape units, and ferrite cores.	12. Storage on magnetic drums, tapes, and ferrite core memory.
13. Magnetic tape store will hold up three days' traffic (57.6×10^6 bits).	13. Magnetic tape store of 16 units. Each will hold 1.6×10^6 words. Total of 25.6×10^6 words or 819.2×10^6 bits. Magnetic drum system of 16,384 words each. Total of 8×10^6 bits. Main Memory (core) of 16,384 words (approximately 0.5×10^6 bits).

Each system has been designed and developed for a specific purpose, and neither appears to be directly applicable to the requirements of the satellite reference network. However, many of the techniques employed in both systems may be considered common to any high-speed store-and-forward system. Some of these applicable techniques include:

- Incoming storage of sufficient capacity to accommodate a maximum length message on each line.
- Overflow storage for peak traffic conditions.
- Outgoing stores which will hold messages in order of precedence when an outgoing line is busy or otherwise unavailable.
- Processing equipment to perform routing and certain record-keeping functions.
- Per-line buffer stores to allow for parity checking, bit rate conversion, and certain identification procedures.

3.5.1 Reference Network Message Center

The terminal facilities represented in the block diagram of Figure 12 are proposed to satisfy the reference network's requirements for a store-and-forward type system. An important feature of this system is the high-speed serial transmission of complete messages on each link. Messages are not frequency-division or time-division multiplexed in the usual sense, except that the store-and-forward traffic time-shares the final bit stream on each link with the real-time traffic. However, only one store-and-forward message is being transmitted at a time on each link. This method minimizes the line equipment needed on each link compared with



2

Figure 12. Typical Store-and-Forward Traffic Functional Diagram for Reference Network Station

that required for several messages being transmitted simultaneously. It also provides greater protection against short interruptions in transmission by affecting a portion of only one message rather than several.

The high-speed bit stream is received on each link and re-clocked. The various channels are demultiplexed (as described in section 3.3) and the store-and-forward channel, as well as the control channel, are routed to the message handling center. The store-and-forward channel consists of 38,400 bits per second, but since they occur only during certain intervals because of the time-division multiplexing, they are actually at a higher rate during those intervals. The 38,400 bits per second allow a variety of possible combinations of data and teletype channels. The maximum number of such channels is:

$$R_T C_T + R_D C_D = 38,400$$

where

R_T = bit rate per teletype channel

C_T = number of teletype channels

R_D = bit rate per data channel

C_D = number of data channels

When the message handling center receives the high-speed, store-and-forward channel, it is in synchronism with the local clock. However, before the messages can be processed or switched, they must be separated and assembled in their entirety. This is accomplished by the use of a magnetic drum. Each incoming link is assigned a section of the drum large enough to store a maximum length message. As each word of the message

arrives in serial form, it must be converted into parallel form and momentarily stored until the proper section of the drum appears under the writing heads. This is accomplished by the use of four-buffer stores per link. The first buffer store accepts eight bits serially. These are then transferred in parallel to the second buffer store. The second store holds 32 bits and is filled by four 8-parallel-bit characters. Its contents are checked for parity. If there are no errors, an acceptance signal is returned to the sending station via the control channel. If an error is found, the contents of the store are discarded, and a request for retransmission of the faulty word is sent to the transmitting station (see Appendix IV). Any other words received between the time of the request for retransmission and the receipt of the retransmission must also be discarded. The start of the retransmission can be signalled by the control channel. The control channel is also monitored for indication of words which should be discarded because they are part of simulated traffic or a transfer interval. When a correct valid word is assembled in the 32-bit buffer store, it is transferred in parallel fashion into a 1,024-bit buffer store. There are two of these per link; one is filling while the other is emptying. The 1,024-bit store is read out, 32 bits at a time, onto the assigned section of the drum. Local teletype and data inputs to the system are handled in much the same manner as satellite links. Each incoming line of 75 to 4,800 bits per second converts the teletype or data code to standard 8-bit Fieldata code (see figure 16). These messages are then assembled in assigned sections of the drum using the same method as previously described. However, the buffer stores can be smaller in size because of the slower bit rate. All links and local

inputs are receiving simultaneously, and the space on the drum must be allotted in such a manner as to permit all the buffer store pairs to unload onto the drum in a single revolution, if necessary.

As soon as the end-of-message indicator has been received, signifying that an entire message has been assembled on the drum, that message is extracted from the drum in parallel form within a single revolution. It is routed to processing if that circuit is free. If another message is still being processed, it is sent to another storage section of the drum or to an overflow tape store from which one message at a time is returned to the drum. When the processing circuit becomes free, it examines the drum storage section for the presence of a message before accepting one from any of the regular assembly sections. If the traffic density should increase to the point where a number of messages were in the overflow tape store awaiting processing, it might prove necessary to return them to the drum in the order of precedence.

All messages entering the processing circuit are concurrently recorded on a master tape machine for a semi-permanent record. Processing consists of examining the heading for a valid addressee, deleting routing indicators in accordance with routing line segregation, and composing multiple-address messages with heading and text for each direction of transmission. Messages found to have an invalid addressee are printed out at the operator's console for manual handling.

During the message processing, the routing information is forwarded to the switching control center. This center applies a bias voltage to the appropriate AND gate to direct the processed message to the desired

output. The message is transferred from the processing circuit through the AND gates to an outgoing tape store, all in 32-bit parallel fashion. One outgoing store is provided per precedence level per link. Two precedence levels are indicated in figure 12. If additional levels are desired, the number of outgoing stores and the size of the switching center must be increased accordingly.

The outgoing stores are unloaded in decreasing order of precedence. Each precedence level tape store must be emptied before the next lower level precedence store can be tapped. Messages are read off the outgoing tape store a word at a time at a speed dependent upon the rate of transmission on the link in which it is to go. Each word is held in a 32-bit buffer store and released at the rate of 8-serial bits per alternate frame for combination with the real-time traffic into a single bit stream.

Messages routed to local outputs are returned by the switching center to drum sections associated with each output line. If the desired section is already in use, the message is directed to an overflow tape store from which it is returned when the space becomes available. From the drum the message is gradually read out into buffer stores to effect speed and parallel-to-serial conversion. Conversion from Fieldata code to teletype or data code must also be made before the message is sent onto the local line.

3.5.3 Handling of a Typical Message

To clarify the handling procedure for teletype and data messages, a sample message will be followed from its origin to its destination.

The source of the sample message is a tributary station connected by land line to a reference network station. At the tributary station the complete message is punched into paper tape using standard 5-bit teletype code. If this had been a data message it would have been punched on standard data cards. Since the trunk is probably a common user one, it may be necessary to wait until the trunk becomes free before transmission of the punched paper tape to the reference network station can begin. If necessary, interruption provisions for high precedence messages can be included. At the reference network end of the trunk, the teletype or data code is converted to standard 8-bit Fieldata code. Buffer stores convert from serial to parallel form and temporarily hold the information block until the drum section assigned to this trunk has rotated into writing position. The entire message is assembled on the magnetic drum in this manner.

As soon as the assembly has been completed, the message is extracted from the drum in a single drum rotation and sent for processing. It is simultaneously recorded on a master tape for semi-permanent file. Following processing, the message is switched to the proper precedence store associated with the desired outgoing link. Here it awaits its turn for transmission. It is read from the outgoing store at a speed dependent on the transmission rate of the link and then enters a 32-bit buffer store which feeds 8-serial bits at a time to the final bit stream under control of the proper time-slot pulses.

The bit stream is transmitted to the other station in the link by radio equipment whose specifications are beyond the scope of this portion

of the study.

At the receiving station, the bit stream is reclocked and demultiplexed as described in the section on real-time traffic. The store-and-forward channel is directed to the store-and-forward center for processing. This channel consists of complete teletype and data messages in serial form. They are not intermixed. One message begins only after the previous one has ended.

Following the progress of the sample message at the received station, it first enters buffer stores for speed and serial-to-parallel conversion. It is then written on an assigned, magnetic drum section and otherwise handled in a manner similar to that at the network station serving the originator. However, if this is the last network station involved in the routing of the sample message, instead of being switched to an outgoing tape store by the switching center, it is returned to the magnetic drum and written in a section associated with the desired local output. From the drum it is gradually read out into buffer stores which serve as speed buffers and parallel-to-serial converters. Following conversion from Fielddata to teletype code, the message is sent on land line to the tributary station where it is punched on paper tape or printed out.

3.6 Equipment Specifications

3.6.1 Storage

Buffer Stores - Buffer stores are required in numerous portions of the reference network terminal equipment for purposes of minor delay as well as serial-parallel and parallel-serial conversions.

All incoming, outgoing, and link traffic is in serial form. To enable more rapid cross-office handling, store-and-forward traffic is converted to 32-bit, parallel form. Since the word thus produced may not occur at the proper portion of the magnetic drum's rotation to be recorded, a buffer store must be provided to delay it as much as one drum rotation, if necessary.

Because of the time-sharing principle used on the links, it may be necessary for a real-time call to change time slots in going from one link to another. The bit rate may also vary from link to link. These changes can be accomplished by holding each character in a buffer store and reading it out in the succeeding frame interval.

The previously mentioned functions of buffer stores require units ranging in size from 4-bits to 1,024-bits (for a drum speed of approximately 3,000 rpm). Any per-bit storage device may be used, such as vacuum tube, gas tube, or transistor flip-flops, magnetic cores, twistors, or tunnel diodes. Buffer stores of this type are well within the state-of-the-art but will probably require a particular design for this application. Transistors may be preferred for the smaller size units, but these become uneconomical in the larger stores where the cores or other new devices become practical.

Magnetic Drum - In the store-and-forward method of message handling, a complete message is usually assembled at a station and stored until traffic conditions and precedence enable it to be transmitted to the next station. This requires some type of large-capacity storage device on each incoming line and link to assemble the complete messages. The most obvious, perhaps,

is magnetic tape. This, however, would require one or two tape machines for each link or local line. A single large magnetic drum could be used instead. This would be more economical and could also serve as the source of all the station timing pulses.

The magnetic drum would be divided into two portions designated "A" and "B". They would have equal storage capacity and would use separate heads.

The "A" portion of the magnetic drum would have to have sufficient capacity to store one maximum length message for each link, incoming line, and outgoing line. In section 3.3.1 of this report the average lengths of military teletype and data messages were shown as 7,000 and 20,000 bits, respectively. If we arbitrarily set a maximum length of 24,576 bits for any type store-and-forward message, an "A" portion drum capacity exceeding one million bits would be required for some network stations. If a longer maximum length message is set, the drum storage capacity must be correspondingly expanded. The "B" portion of the drum would permit storage of completed messages which had been removed from the "A" portion to permit reception of a new message but had been unable to enter the processing circuit because of a busy condition.

To permit rapid cross-office handling of traffic, information would be stored on the magnetic drum in parallel. Thirty-two bits would be written or read simultaneously. The assignment of drum sections in the "A" portion to the "N" lines and links must be done in such a manner that they can all be scanned in one drum revolution. They would be intermixed so that every Nth bit on each track would be associated with the same line or link. In

the "B" portion, messages would occupy an available section without specific assignment, but the same type of intermixing would be used. For a drum with an "A" portion capacity of 1,000,000 bits, and a speed of 3,000 rpm, this would result in a cross-office speed of approximately 40,000-bits per second on each of the 32-leads. Stations with fewer inputs could have the cross-office speed correspondingly reduced.

Phonic discs or clock tracks on the magnetic drum would be used as the source of all timing pulses at a reference network station. In the worst case, one such disc or track would be required for each bit rate encountered at the station. However, by the use of bit rates which are multiples of one another, the number of basic sources required could be reduced.

Magnetic drums of the type required for this application are in common use today and should require no special development.

Output - The store-and-forward traffic consisting of data and teletype messages must have storage facilities at the output of each link. This is due primarily to the fact that the cross-office handling rate is much faster than the link transmission rate which may cause a backlog of messages to build up awaiting transmission on a particular link. Selection of messages for transmission in order of precedence, periodic discontinuities in transmission for purposes of satellite transfer, as well as occasional lack of satellite coverage, are additional reasons for needing output stores.

Since the cross-office handling is done in parallel form, the output stores must be capable of handling 32-bits simultaneously. They must be able to read in at a fast speed determined by the cross-office rate

and read out intermittently at a slower speed determined by the store-and-forward channel bit rate. A loop-type, magnetic tape machine has been selected as the most suitable for this purpose. It must have 32 information tracks and two sets of heads, one set for writing and one set for reading. Different speed, separate drive mechanisms are required to move the tape past each head. Slack tape is allowed to accumulate between the sets of heads. The product of the tape-packing density and the tape-writing speed must be equal to the cross-office handling rate. Similarly, the product of the tape-packing density and the tape-reading speed must be equal to $\frac{38,400}{32}$ or 1,200 bits per second. The length of the tape loop is a function of the traffic distribution according to destination and precedence, the traffic load, and the satellite outage time. In most cases, it will probably best be determined experimentally.

To permit the transmission of messages in the order of precedence, a separate tape machine is required on each link for each level of precedence recognized by the system. All messages of equal precedence destined for the same station are recorded on the same tape machine. Read-out is in order of precedence, followed by chronological order within each precedence. If a high precedence message arrives during transmission of a lower precedence message, pre-emption of the channel by the high precedence message may occur immediately, or its transmission may follow that of the current low precedence one. At the end of every message transmission, each output tape machine on the link is scanned for the presence of a message, and the highest precedence machine having one ready is accepted next.

No currently-available, loop magnetic tape machine is known which

meets all the requirements of this application. It must stop its read out for approximately 30 milliseconds every minute to provide a satellite transfer interval. This requires a start-stop time of less than one millisecond or the ability to stop and back up slightly during the transfer interval to avoid the loss of any information. For some stations it may need to accept an input of 40,000 bits per second and produce an output of 1,200 bits per second on each of 32 tracks. These features will necessitate some development effort.

Overflow - Since the space available on the magnetic drum is necessarily rather limited, some storage facilities must be provided for messages which attempt to enter a portion of the drum and find it already occupied. This will occur principally in two cases. The first is in the assembly of incoming messages. Each message can, if necessary, be followed immediately by another one. This means that it must be cleared from the drum promptly to provide storage space for the following message. If the processing circuit is free, the message is directed there. However, if it is in use, the message must enter the other portion of the drum to await the availability of the processing circuit. If this portion of the drum becomes full, some messages must be removed to an overflow store.

The second case requiring the use of an overflow store is the distribution of local outputs. Each output line is assigned a drum section from which a message is gradually read out. If another message arrives for this drum section while it is occupied, it must be diverted to an overflow store until the previous message has been completely read out.

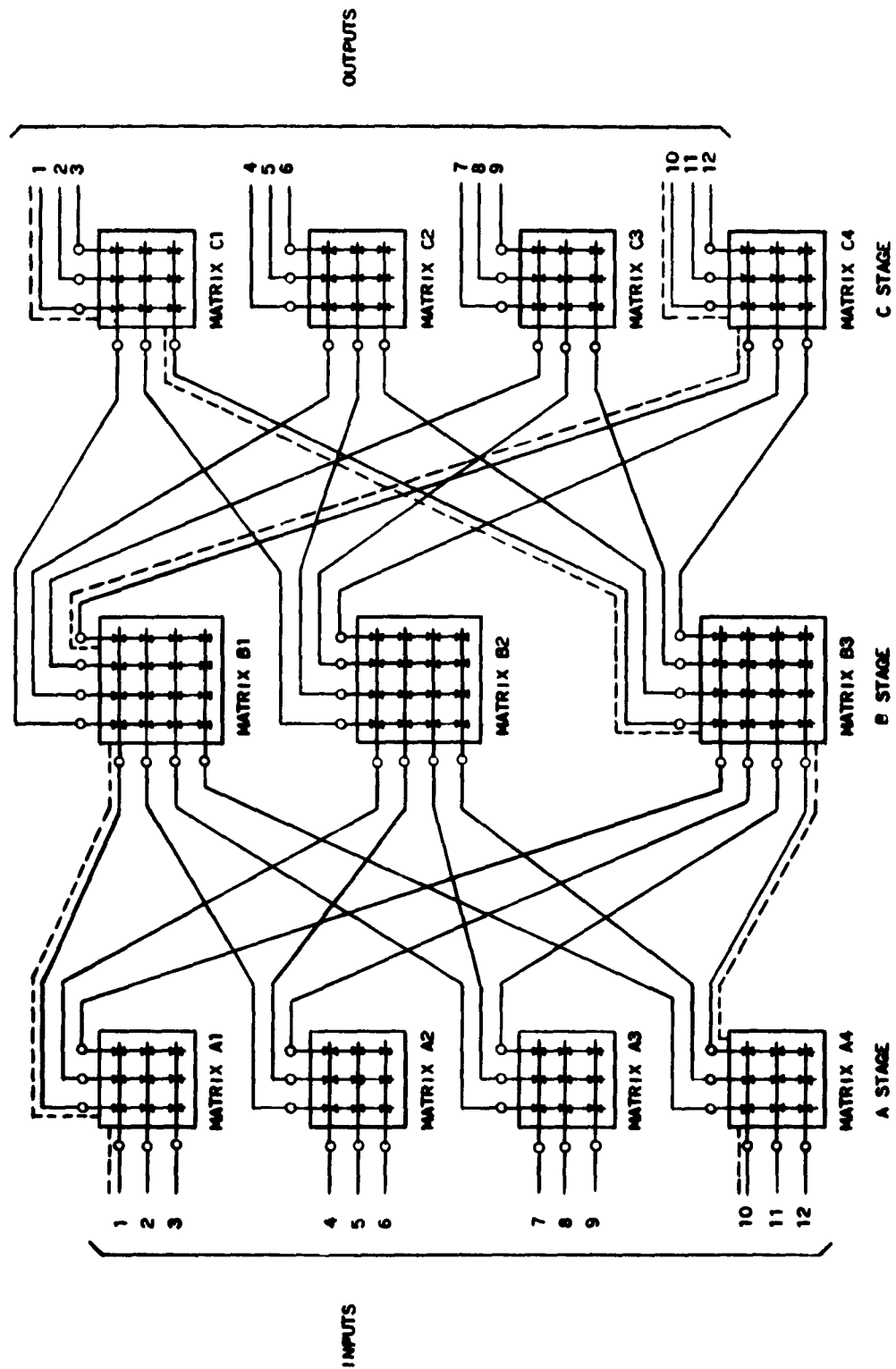
A reel-type, tape machine should prove satisfactory in each of these cases. However, it must be a high-speed one with 32 information tracks capable of exchanging information directly with the drum at a rate of approximately 40,000 bits per second on each track. This may require some development work.

3.6.2 Switching

A non-blocking space-division, switching array is provided in the circuit-switching center for the reasons discussed in section 3.4.2. This uses multiple stages of selection, and sealed, precious metal, pressure contact crosspoints to provide a continuous metallic path which is utilized only during the periods designated by control pulses. Connections through the array are established in 10-20 milliseconds on a one-at-a-time basis under the control of electronic common equipment. This method simplifies the design and maintenance of the common control equipment. The high speed of operation permits the use of electronic control, thus allowing standby equipment to be continuously routined. In this manner, a fault in standby equipment can be found and cleared before the equipment is needed for service. A simplified switching array is shown in figure 13. A connection set up through the array consists of 16 wires in the real-time switching center (4-wires for duplex signal transmission and 12-wires for timing pulses to control read-out from the buffer stores).

Dry reed relays were selected as crosspoints for the following reasons:

- a. High switching speed (approximately 1 millisecond).
- b. Wide bandwidth.



NOTES:

1. PATH DRAWN IN THIS - - - SHOWS TWO OF THE POSSIBLE WAYS TO CONNECT CIRCUITS 1 AND 10.
2. - - - REPRESENTS A CROSSPOINT

Figure 13. Simplified Switching Array

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- c. Low noise and crosstalk.
- d. Small delay distortion.
- e. Less maintenance and greater reliability than any other currently available switching system.
- f. Does not require controlled environment.
- g. Readily modified for system expansion.
- h. No adjustments necessary.
- i. Long life.

A typical, dry reed relay switching center is shown in figure 14. The incoming lines terminate through the trunk circuits on the non-blocking, three-stage, switching matrix. The switching matrix provides for setting up the connection between the calling and the called terminals. The trunk circuits provide signal conversion and supervision to the common equipment. The common equipment (scanner, marker-translator, path selector, registers, and senders) supervises and initiates the connections through the switching matrix. The scanner continuously monitors the terminals for requests for service, if such a request is encountered, it transmits this information to the marker. The marker, in connection with the static translation field, recognizes the grade of service required and correspondingly calls in the path selector and a free register to initiate the connection of the calling terminal to the register through the switching matrix. A further function of the marker is to assist registers, senders, and the path selector in setting up and supervising connections. The path selector controls the switching matrix and sets up the crosspoint connection. The registers are connected to the calling terminals and store the address information received

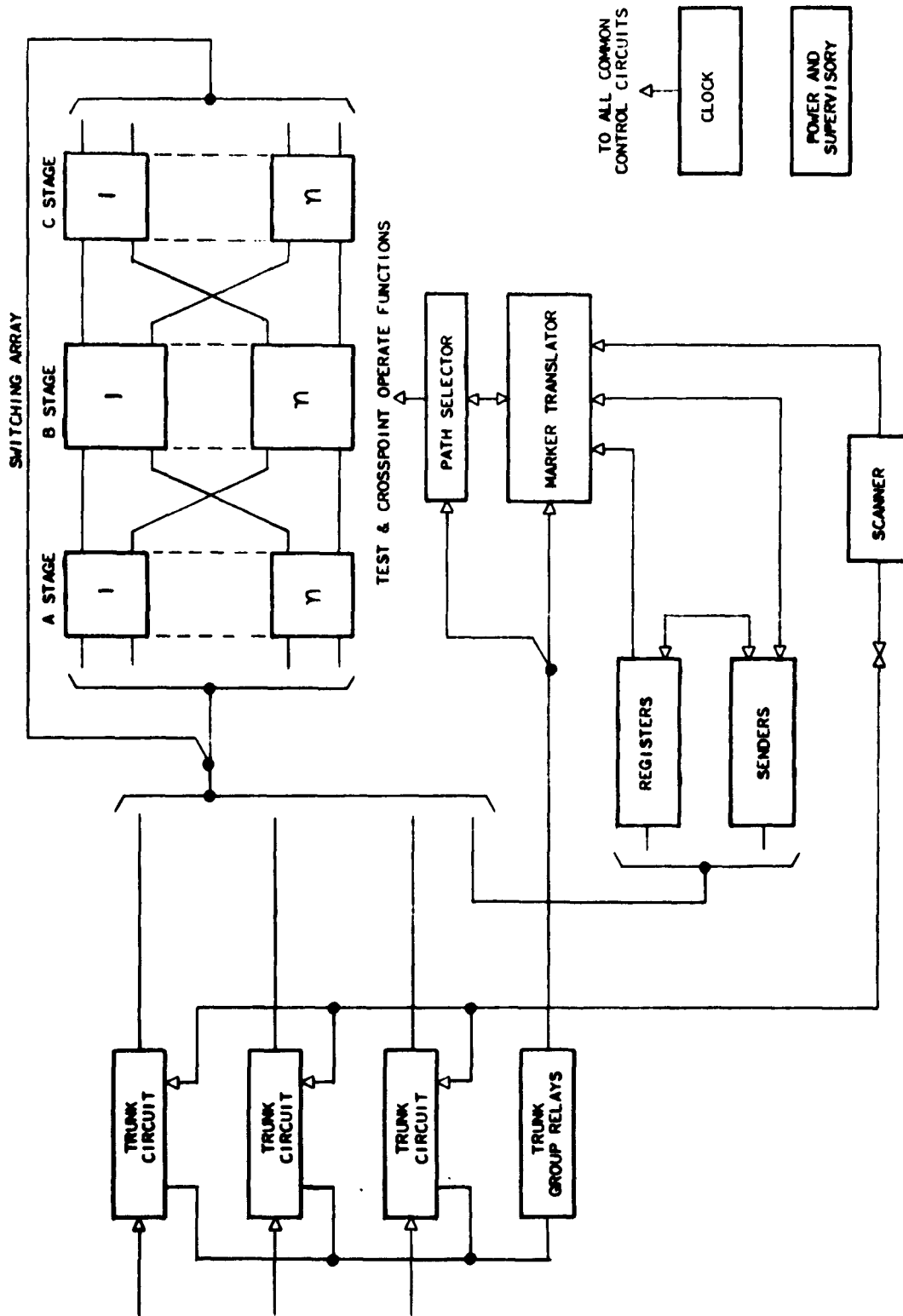


Figure 14. Typical Reed Relay Switching Center

from the line or trunk. The senders, in connection with the registers, control the outgoing signalling and inter-office transfer of routing information.

Dry reed relay, switching centers of the type described here are compatible with the state-of-the-art.

3.6.3 Multiplexing and Demultiplexing

The two principal types of multiplexing, time-division and frequency-division, are described briefly in section 3.3.2. Although both are proposed for use in the reference network, the one of primary concern to the terminal equipment is time-division multiplexing.

Time-division multiplexing is used to combine into a single bit stream all the traffic on each link. This is accomplished electronically by means of master clock wheels included in the magnetic drum which serve as the source of all timing pulses used at each station. Each channel on a link is allotted a time interval by means of these pulses. Two channels on different links may use the same or overlapping time intervals but, within a single link, each channel must have its own time slot.

The information to be sent on each outgoing channel is stored in parallel form in character buffer stores. The size of these stores is dependent on the number of bits per character in the particular type of traffic; i.e., four bits for voice, six bits for graphics, and eight bits for teletype and data. When any of the voice or graphics stores is connected to an outgoing channel by the switching center, the assigned time slot pulses for that channel are also connected into the output of the character buffer stores. Each teletype and data output store is tied

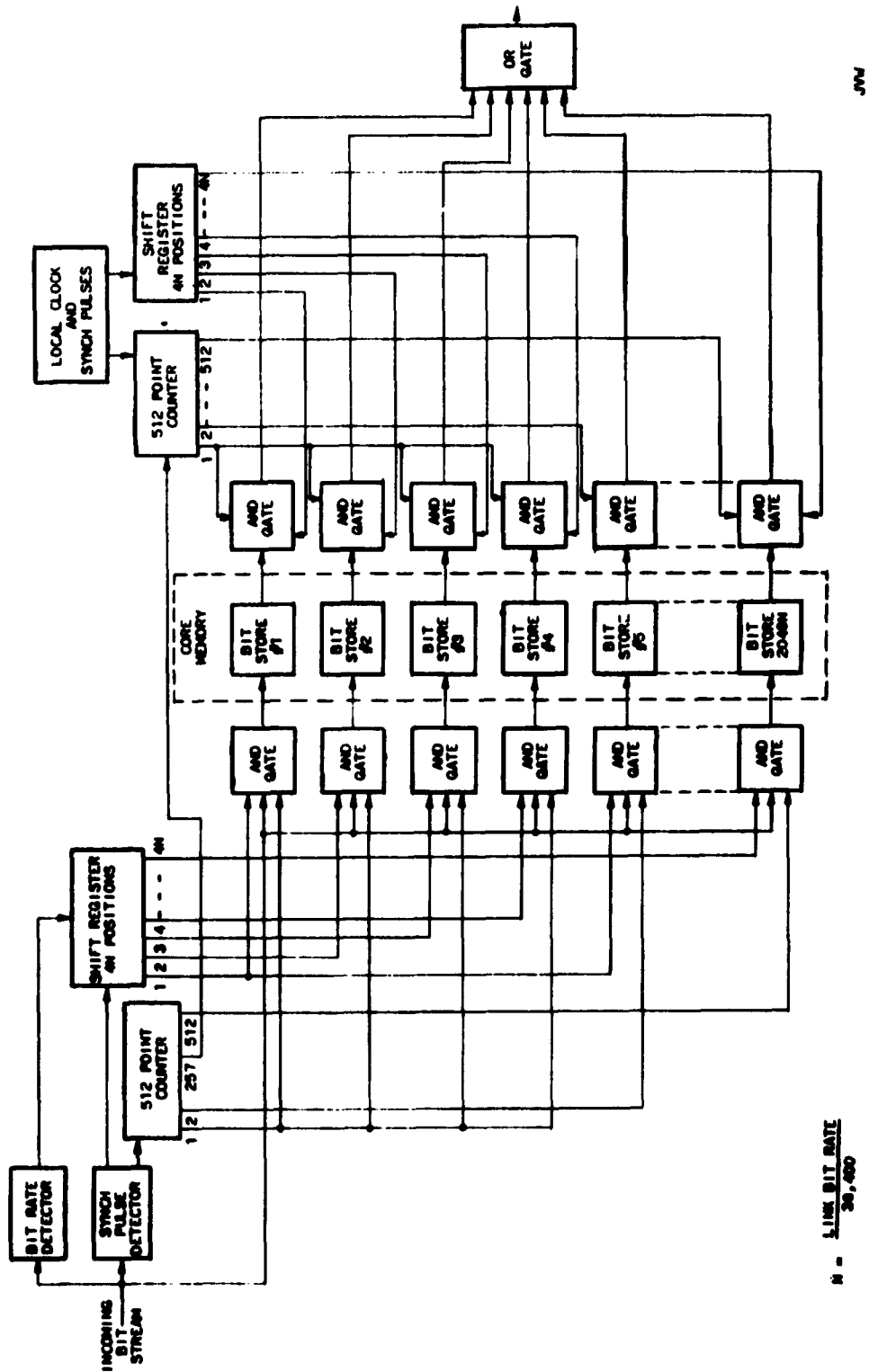
directly to a multiplexer, and the necessary time slot pulses are permanently connected. In this manner, the read-out from the buffer stores is controlled by, and occurs only during, the proper time intervals. The various bits of information, each one having its own unique time slot on a particular link, are then combined to form a continuous binary stream. Therefore, the multiplexer, in this case, is basically nothing more than a large "OR" gate.

In a comparable fashion, the binary streams being received from other stations must be separated into their individual channels. Since they were originally combined by other clocks not in synchronism with the local station clock, all incoming binary streams must be reclocked (section 3.6.4) to get them in step with the local clock and with each other. The reclocked bit stream is gated with each of the appropriate channel, time-slot pulses from the local station clock. Each voice and graphics channel is routed onto a separate line in preparation for circuit switching. The combined teletype and data channel is routed to the message switching center.

3.6.4 Reclocking

The bit streams received by a reference network station must be reclocked to compensate for Doppler effect caused by satellite movement and to synchronize the traffic received from various non-synchronous stations. All incoming traffic is thereby brought into synchronism with the local station clock enabling demultiplexing, switching, and remultiplexing to easily take place.

The reclocking process as shown in figure 15 consists of continuously reading into a magnetic core storage unit in synchronism with the distant transmitting station clock by means of timing pulses inserted in



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N = $\frac{\text{LINK BIT RATE}}{30,400}$

Figure 15. Reclocking Block Diagram

the bit stream, and continuously reading out in synchronism with the local station clock. As discussed in section 3.3.3, a framing pulse is inserted at regular intervals of $1/9,600$ second. This may be recognized either by its uniqueness or by its repetition rate. The number of information bit intervals between successive framing pulses is dependent upon the number of channels in the link. The framing pulses are detected and shifted along through a shift register at a rate determined by the frequency of the bit stream. The length of the shift register is equal to the number of bit positions in a framing interval. The output of each shift register stage is combined in a separate AND gate with the incoming bit stream to direct each bit to a separate position in a magnetic core store.

For satellites at an altitude of 2,200 miles, the variation between maximum and minimum transmission path length is approximately 4,400 miles. This is equivalent in propagation time to 23.7 milliseconds. To permit retiming of the bit stream over this maximum variation, the magnetic core store must be capable of holding the number of bits which can be received during this time interval. The link bit rate is defined as:

$$\text{Link bit rate} = 38,400 \times N$$

Where N is the number of 38,400 bit per second channels required for handling the traffic on a link.

A link operating at $38,400 \times N$ bits per second will receive $910 \times N$ bits during a 23.7 millisecond period. It must also be able to compensate for any change in the relative speeds of the clocks at the two stations. If atomic standards are used for periodic reference,

crystal oscillators can be set to 2×10^{-10} , and aging and environmental effects on the crystals can be controlled so as to keep the total absolute frequency error below 1×10^{-8} for a period of ten or more days. If it is assumed that a weekly clock correction must be made, assuming a frequency error of 1×10^{-8} the clock drift for a bit rate of $38,400 \times N$ bits per second for one week will amount to only $231 \times N$ bits.

$$\text{Clock Drift} = 38,400 \times N \frac{\text{bits}}{\text{second}} \times \frac{\text{frequency error}}{\text{error}} \times 7 \times 86,400 \frac{\text{seconds}}{24 \text{ hrs.}}$$

or excess storage capacity required

$$\text{Clock Drift} = 38,400 \times N \times 1 \times 10^{-8} \times 86,400 \times 7$$

$$\text{Clock Drift} = 231 \times N \frac{\text{bits}}{\text{week}}$$

Therefore, for a link operating at $38,400 \times N$ bits per second, a storage capacity of $2048 \times N$ bits (which is the next larger power of two) will fulfill these requirements.

The requirement of resetting the local clocks may be accomplished in the following manner. It is possible to achieve a measure of frequency error relative to all of the adjacent stations by sampling the buffer capacity and averaging. By systematically starting from the 21 gateway station (one only) and proceeding outward on the link to successive A stations, then to their associated B stations, and finally to the C stations that a system calibration can be effected. This calibration can be done while handling traffic; therefore, it should not compromise the operating reliability of the system. For a more detailed analysis of this problem refer to Apprndix V. In this manner, a maximum path length variation in either direction can be tolerated.

Since the capacity of the magnetic core store is equal to 512

framing pulse intervals, the shift register must complete 512 cycles to entirely fill the store once. This requires a 512-point counter to determine which cycle the shift register is in and to route the incoming bits to the proper store.

The procedure for read-out from the core storage unit is essentially the inverse of the read-in procedure, with the difference that the output shift register is controlled entirely by the local clock, and the output 512-point counter has its driving pulses gated with the input 512-point counter in such a manner as to prevent the output counter from starting until the input counter reaches the center position. Once having started, the output counter is independent of the input counter.

The complete reclocking process shown in figure 15 is not known to be commercially available. However, the individual circuits comprising the reclocking function are either available or are consistent with the state of the art and could be produced. Examples of commercially available circuits similar to those required for this purpose are cited:

AND gates, OR gates, counters, shift registers --

Engineered Electronics Company, T series packaged circuits;
ACF Electronics Division, Digital Logic Module System;
CHK Components Inc., encapsulated modules;
Raytheon Company Semiconductor Division, Circuit-Paks;
Digital Equipment Corporation, System Building Blocks;
Computer Control Company, Inc., Series S Pacs;
Rese Engineering, Inc., Logix Blocks.

Magnetic Core Memory --

Computer Control Company, Inc.;
General Ceramics, Applied Logics Dept.;
Burroughs Corp.;
Rese Engineering, Inc.

3.6.5 Code Conversion

All teletype and data traffic entering the reference network will be converted from its own code to 8-bit Fieldata code (see figure 16). This is a standard military code with six information bits, one parity bit, and one control bit. The use of Fieldata will provide for system uniformity and error checking of store-and-forward traffic. One converter may be used for each input device, or for a common user system, one converter may be used for each input line to the reference network. The latter method requires that data and teletype inputs be segregated on separate lines or that a feature be included to distinguish between the types of traffic. In either case, the converted character is read out in parallel form to the buffer store.

Information leaving the reference network will pass through converters whose function is the inverse of those just described; i.e., they will convert from Fieldata code to teletype or data code.

Code converters of this type are relatively common and will require no research or development.

3.6.6 Analog-Digital Conversion

The exclusive use of digital signals in the reference network necessitates the conversion of analog voice and graphics information into binary coded form. This is accomplished by sampling the amplitude of the analog signal at a rate at least twice the highest frequency occurring within it. The amplitudes thus obtained are converted to the nearest binary digit possible in the selected code.

However, since the amplitude change between adjacent samples is usually small, a fewer number of bits can be used to represent the

7 6 5 4 3 2 1 0
P C 1 2 3 0 1 0

ALPHANUMERIC										CONTROL																
CHAR- ACTER	7	6	5	4	3	2	1	0	CHAR- ACTER	7	6	5	4	3	2	1	0	CHAR- ACTER	7	6	5	4	3	2	1	0
MASTER SP.	0	1	0	0	0	0	0	0)	1	1	0	0	0	0	0	0	BLANK/IDLE	1	0	0	0	0	0	0	0
UPPER CASE	1	1	0	0	0	0	1	0	+	0	1	1	0	0	0	0	1	TST	0	0	0	0	0	0	0	1
LOWER CASE	1	1	0	0	0	0	1	0	=	0	1	1	0	0	0	0	1	TCL	0	0	0	0	0	0	0	1
LINE FEED	0	1	0	0	0	0	1	1	>	1	1	1	0	0	0	1	1	TAB	1	0	0	0	0	0	1	1
CARRIAGE RE- TURN	1	1	0	0	0	1	0	0	<	1	1	1	0	0	1	0	0	CONTROL CR	0	0	0	0	0	1	0	0
SPACE	0	1	0	0	0	1	1	0	*	1	1	1	0	0	1	1	0	CONTROL SPA.	1	0	0	0	0	1	0	1
A	1	1	0	0	0	1	1	0	!	1	1	1	0	0	1	1	0	CONTROL A	0	0	0	0	0	1	1	0
B	1	1	0	0	0	1	1	0	"	1	1	1	0	0	1	1	0	CONTROL B	0	0	0	0	0	1	1	0
C	1	1	0	0	0	1	1	0	#	1	1	1	0	0	1	1	0	CONTROL C	0	0	0	0	0	1	1	0
D	0	1	0	0	1	0	0	1	%	1	1	1	0	0	1	0	1	CONTROL D	0	0	0	0	1	0	0	1
E	0	1	0	0	1	0	1	0	&	1	1	1	0	0	1	0	1	CONTROL E	0	0	0	0	1	0	1	0
F	0	1	0	0	1	0	1	1	'	1	1	1	0	0	1	0	1	CONTROL F	0	0	0	0	1	0	1	1
G	1	1	0	0	1	0	1	1	(1	1	1	0	1	1	0	0	CONTROL G	1	0	0	0	1	1	0	0
H	1	1	0	0	1	1	0	1)	1	1	1	0	1	1	1	0	CONTROL H	0	0	0	0	1	1	0	1
I	1	1	0	0	1	1	1	0	*	1	1	1	0	1	1	1	0	CONTROL I	0	0	0	0	1	1	1	0
J	1	1	0	0	1	1	1	1	=	1	1	1	0	0	0	0	0	CONTROL J	0	0	0	1	0	0	0	0
K	1	1	0	1	0	0	0	1	>	1	1	1	0	0	0	1	0	CONTROL K	0	0	0	1	0	0	0	1
L	0	1	0	1	0	0	1	1	<	1	1	1	0	0	1	0	1	CONTROL L	1	0	0	1	0	0	0	1
M	0	1	0	1	0	0	1	1	%	1	1	1	0	0	1	1	0	CONTROL M	0	0	1	0	0	1	0	1
N	0	1	0	1	0	0	1	1	'	1	1	1	0	0	1	1	0	CONTROL N	0	0	1	0	0	1	0	1
O	1	1	0	1	0	0	1	1	(1	1	1	0	1	0	1	0	CONTROL O	1	0	0	1	0	1	0	1
P	1	1	0	1	0	1	0	1)	1	1	1	0	1	0	1	0	CONTROL P	0	0	0	1	0	1	0	1
Q	1	1	0	1	0	1	0	1	*	1	1	1	0	1	0	1	0	CONTROL Q	0	0	0	1	0	1	0	1
R	1	1	0	1	0	1	1	0	>	1	1	1	0	1	1	1	0	CONTROL R	0	0	0	1	0	1	1	1
S	0	1	0	1	0	1	1	0	<	1	1	1	0	0	0	1	0	CONTROL S	1	0	0	1	0	1	1	0
T	0	1	0	1	0	1	1	0	%	1	1	1	0	0	0	1	0	CONTROL T	0	0	0	1	1	0	0	1
U	1	1	0	1	0	0	0	1	'	1	1	1	0	0	1	0	1	CONTROL U	0	0	0	1	1	0	0	1
V	1	1	0	1	0	0	1	1	(1	1	1	0	0	1	0	1	CONTROL V	0	0	0	1	1	0	0	1
W	1	1	0	1	0	1	0	1)	1	1	1	0	1	1	1	0	CONTROL W	0	0	0	1	1	1	0	0
X	0	1	0	1	0	1	1	0	*	1	1	1	0	1	1	1	0	CONTROL X	1	0	0	1	1	1	0	1
Y	0	1	0	1	1	0	1	1	=	1	1	1	1	1	1	1	0	CONTROL Y	1	0	0	1	1	1	1	0
Z	1	1	0	1	1	1	1	1	>	0	1	1	1	1	1	1	1	CONTROL Z	0	0	0	1	1	1	1	1
									SPECIAL IDLE																	

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Figure 16. Fieldata Standard Code Table

amplitude variation rather than the absolute amplitude. Such a method is known as differential pulse code modulation (PCM).

The characteristics of differential PCM make it well suited for the digitization of speech. Since voice frequencies extend to slightly above 4,000 cycles per second, a sampling rate of 9,600 samples per second will be sufficient. The variation in amplitude of these samples will then be converted into digital form using a 4-bit binary code. This will provide a relatively high quality digital speech signal of 38,400 bits per second, which is in keeping with the system of 75×2^N bits per second which seems most likely to be adopted by the Committee on Military Communication System Technical Standards.

Analog facsimile signals do not contain as high frequencies as analog voice signals; therefore, the sampling rate for digital conversion of analog facsimile can be reduced to 1,600 samples per second. A six-bit binary code results in a bit rate of 9,600 bits per second also in the form of 75×2^N . In the present state of the art, a six-bit code is considered the minimum necessary to produce a picture with barely detectable distortion when reproduced. If a higher sampling rate should become necessary because of more rapid scanning, this can easily be accommodated as long as the resulting bit rate multiplied by $4/3$ is an integral subdivision of 38,400. This can be seen by referring to section 3.3.3 where the graphics channels are shown alternating with the store-and-forward channel which occupies 8 bits. Therefore, 8 bits are available for the graphics channels even though they only require 6 bits.

Each trunk entering the reference network will require either a voice or facsimile converter. Analog-digital converters of the type

proposed here are consistent with the present state of the art. Although not known to be commercially available exactly as needed, they should require little or no development work. A functional block diagram³⁵ for a possible voice differential PCM encoder is shown in figure 17. This operates in the following manner. The analog voice signal enters the subtractor at the point marked IN. Here it is compared to the integrated previously quantized value. The difference signal is amplitude-sampled and quantized to one of 16 possible values. The selected value is converted to a four-digit code. The quantizing scale shown here is appropriate for a toll telephone system. It may be possible in the reference network to decrease these values for better quality if transmission variations can be held sufficiently low.

Analog facsimile signals can be converted to six-bit PCM by means of a circuit³⁵ similar to the one shown in figure 18. Here the input signal is shown combined with the output of a binary network at the input to a comparator or decision-making circuit. An orderly succession of binary comparisons are made starting with the largest value. The positive or negative output of the comparator for each comparison gives the serially coded signal directly.

3.6.7 Message Processing

Messages are removed, one at a time, from the magnetic drum system in 32-bit parallel fashion for processing. This involves several steps performed in rapid sequence. The heading is examined for a valid

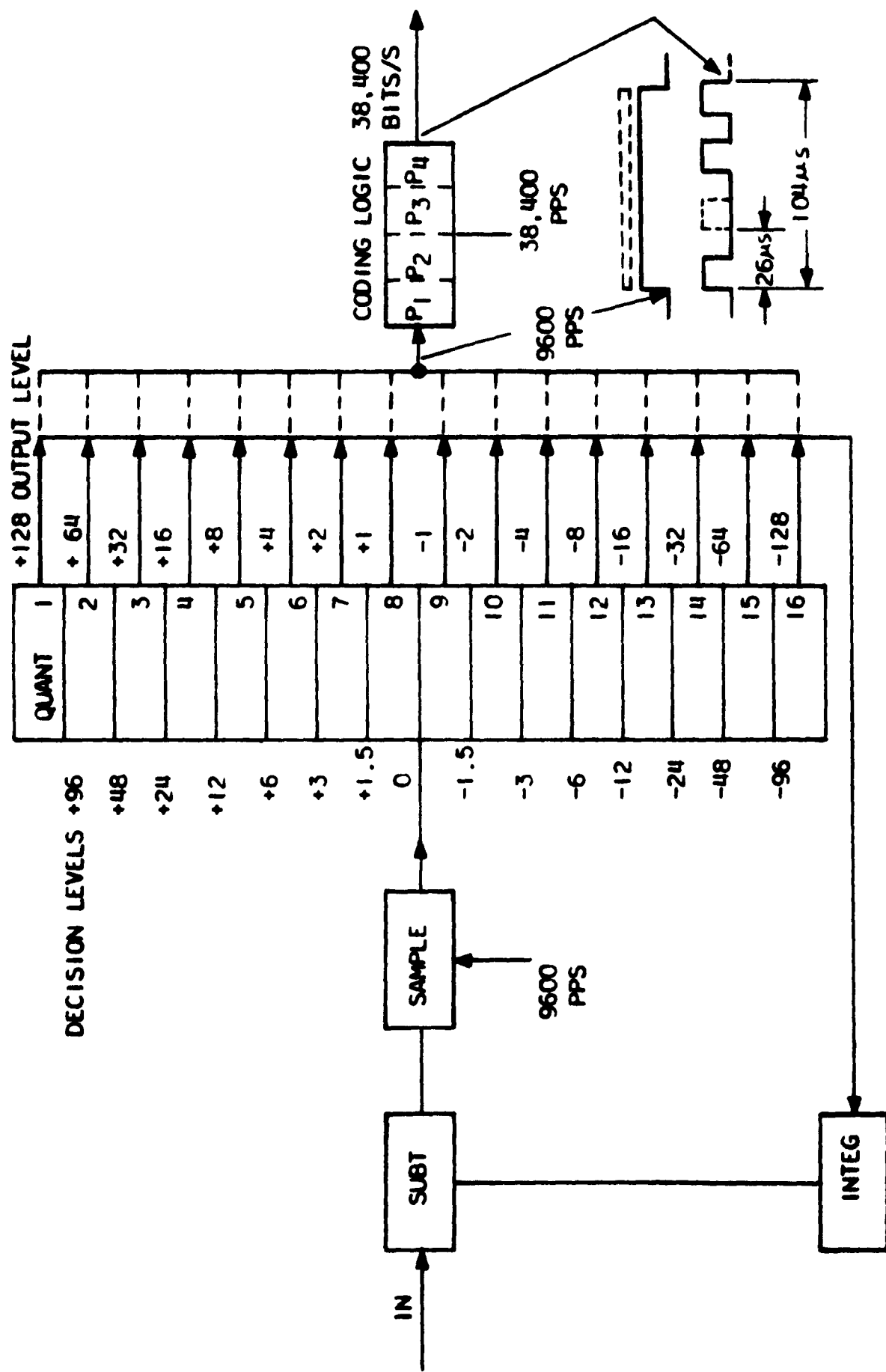


Figure 17. Voice Differential Encoder

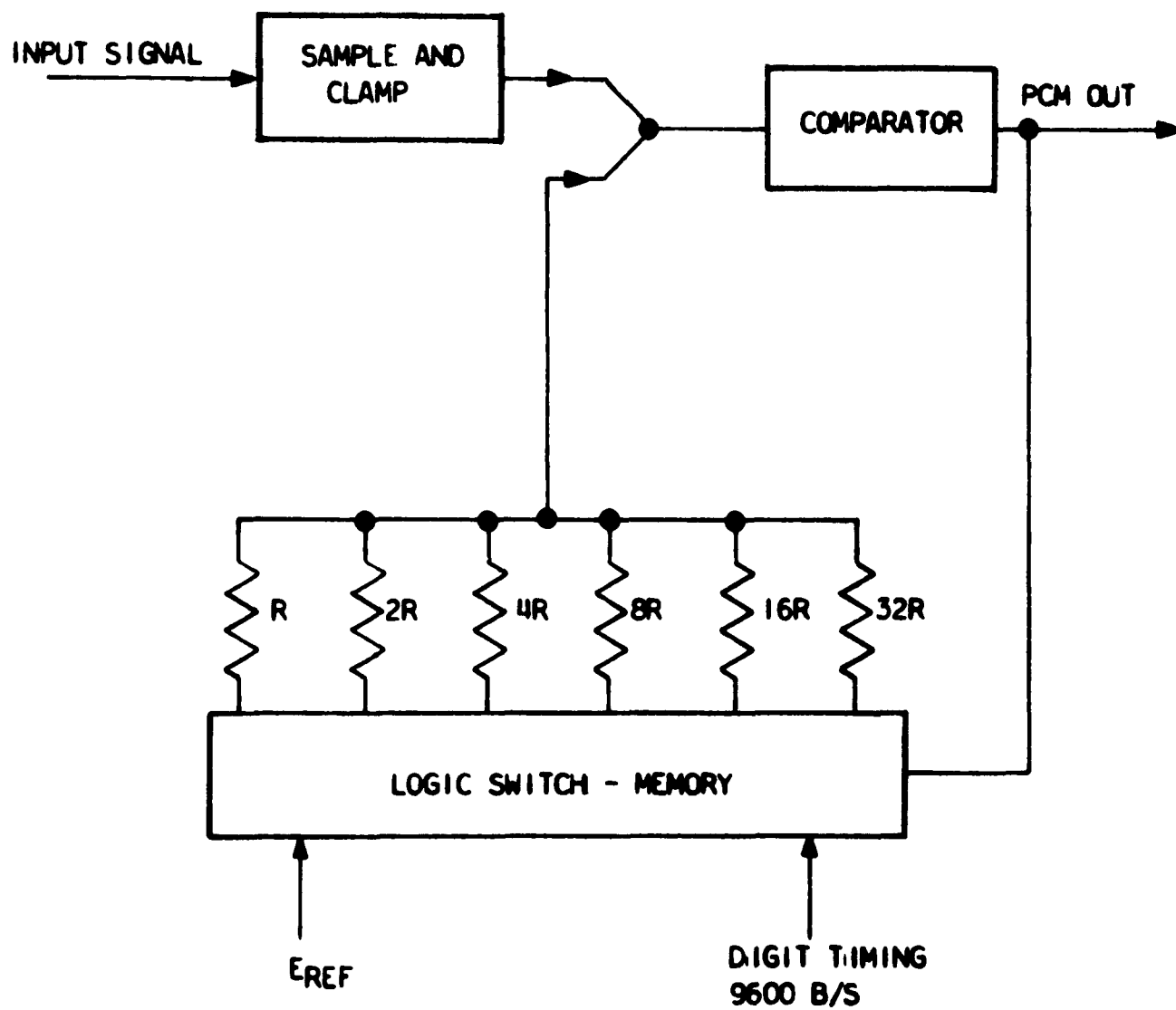


Figure 18. Facsimile PCM Encoder

addressee. Routing indicators for this station are deleted from the message to be forwarded. When the message must be sent on more than one link, a heading is composed for each transmission and tied to the text of the message. If any of these operations cannot be performed in the usual automatic manner, the operator is alerted by a signal on the console to manually intervene and correct the difficulty. At the completion of processing, the messages are forwarded in 32-bit parallel fashion to the proper output.

The message processing requirements of the reference network do not necessitate the complexity of a multisequence computer such as the SPE of the SAC Control System. The EDTCC must perform other functions as well as message switching. For example, it must keep on record, and continually update, a library of information which must be available on call to the SAC control operators. Because of the less complex demands on the reference network message processing facilities, simpler wired logic should prove sufficient.

A computer capable of performing the operations necessary for message processing in the reference network is within the present state-of-the-art.

3.7 Equipment Pricing

3.7.1 Pricing Basis

The ground station configurations illustrated in figures 11 and 12 will apply basically to all sizes of ground stations encountered in the reference network. The actual complement requires will differ only in the amount of line equipment deemed necessary by the individual

needs of the stations. Thus, the building-block approach may be used in specifying the terminal equipment requirements.

Both the real-time (figure 11) and store-and-forward (figure 12) portions of the terminal equipment required at the ground station have common equipment that is required regardless of the number of local lines and satellite links needed on a station basis. Some variation in size of the common equipment can be tolerated with regard to the ultimate capacity of the station; however, the difference in cost reflected here is small compared to that of individual line equipment. At this time this factor will be taken into consideration only for the store-and-forward common equipment.

In the study, no pattern for the number of local lines required at various ground stations could be found. This means that a type "C" station can have as many, or more, local inputs and outputs as a type "A" station. In trunking capacity for communication via satellite where type "A" stations generally handle the heaviest amount of traffic, the situation is more predictable. However, many type "B" stations equal, and in one instance better, the type "A" stations in trunking capacity required. Therefore, from the pricing point of view, there are large stations and small stations.

The real-time switching system (figure 11) is to a large extent, expandable on a per-line basis. Common equipment required is the reed relay switching center control, the voice and graphics channels, traffic simulator, and the control and clock pulse circuit. In the store-and-forward (figure 12) portion of the ground station con-

figuration, a larger percentage of the equipment required is common. Chief among this equipment is the magnetic drum system. It is recommended that drums of different capacity be used for the small and large size stations. Two basic models will be sufficient. The remaining common equipment is essentially the same whether the station is small or large in size.

Proposing a large scale system of this sort always brings forth the question of price. The equipment techniques suggested in this system are perhaps three to five years in the future for some items. The prices given below are judicious estimates of the purchase price that would have to be paid at that time to implement the system hardware. They are based chiefly upon ITT Laboratories' experience with such communication and data handling projects as the SAC 465L System, the Binary Information Exchange (BIX), the Automatic Message Recording System (AMR) for electronically billing toll telephone calls, a PCM voice communication system developed for the U. S. Army Signal Corps, and several multiplexing equipments, (e.g. AN/TCC-13, AN/TCC-15, AN/TCA-2). No development charges are included in the pricing structure. Below, each of the equipment items found on figures 11 and 12 are priced along with a total amount for individual and line equipment.

3.7.2 Pricing List

Real Time System (figure 11)

<u>Equipment</u>	<u>Price</u>
Analog-Digital Converter	\$5,000
Buffer Store - 4 Bit	100
Buffer Store - 6 Bit	150
Control & Clock Pulses Ckt	10,000
Digital-to-Analog Converter	5,000
Demultiplexer	500/line output
Multiplexer	500/line input
Reclock Ckt	10,000
Reed Relay Switching Center)	
Reed Relay Switching Center)	1,000/line
Reed Relay Switching Center Control)	
Voice & Graphics Channels Traffic Simulator	10,000
Common Equipment - Total Price	20,000
Local Duplex Line Equipment - Voice or Graphics	
Total Price Per Line	11,200
Satellite Link Duplex Equipment - Total Price	
Per Link	12,300

NOTE: - The cost of the terminal equipment at a reference network station can be estimated by combining the proper multiples of the above prices. For the real-time station shown in figure 11, the total terminal equipment cost would be:

Common Equipment	\$20,000
10 Local Duplex Lines - 10 x \$11,200 =	112,000
2 Duplex Satellite Links - 2 x 12,300	24,600
	<u>\$156,600</u>

Total Price of Real-Time Terminal Equipment Shown in figure 11.

Store-and-Forward System (figure 12)

<u>Equipment</u>	<u>Price</u>
Buffer Store - 8 Bit	\$ 200
Buffer Store - 32 Bit	1,000
Buffer Store - 256 Bit	5,000
Buffer Store - 1,024 Bit	15,000
Code Converter	200
Control & Clock Pulses Ckt	10,000
Magnetic Drum System - Large Size Station	50,000
Magnetic Drum System - Small Size Station	30,000
Master Tape Storage	40,000
Operator's Console	100,000
Overflow Tape Storage	40,000
Output Store	50,000
Processing	500,000
Output Gating	6,000/32 line output
Switching Control	10,000
TTY and Data Traffic Simulator	10,000
Common Equipment - Small Stations - Total Price	790,000
Common Equipment - Large Stations - Total Price	810,000
Local Duplex Line Equipment - TTY - Total Price	
Per Line	2,400
Local Duplex Line Equipment - Data - Total Price	
Per Line	10,200
Satellite Link Duplex Equipment - Total Price	
Per Link	132,400

Following the procedure outlined above for the real-time system, the total price of Store-and-Forward terminal equipment shown in figure 12 is \$1,084,800.

4. CONCLUSIONS

One phase of this program has been the continuation of the original traffic study, which has been described in an earlier report.² A recommendation of that report was the obtaining of actual traffic statistics that could be used to predict the traffic flow through the network during the 1965-1970 period. Since such information has not been available, it has been necessary to retain the original reference network with relative trunking capacities determined by the comneeds. However, the traffic study has provided the general information determining the routing, precedence, message lengths, subscriber needs, and other traffic characteristics as they would apply to this particular network. From these characteristics, and those of the orbital system, a specification for terminal equipment has been given in terms of a representative station, rather than for each specific station in the network.

In determining the terminal equipment specifications, a major problem area has been the need for handling all types of traffic. A study of the traffic characteristics in present Air Force communications networks also indicates that no one global network can handle the total volume of Air Force traffic. The satellite network appears best suited as a common user system, handling basically routine information. This would reduce the load on other transmission media and would make available more channel space for tactical and "hot-line" information.

The conclusions of this study may be summarized as follows:

1. Reference network should be a common user system.
2. Network terminal is a combined switching center, handling both real-time and store-and-forward traffic.

3. Analog signals to be converted to digital.
4. Outputs to be time-division multiplexed into a single binary stream.
5. Multi-link outputs to be frequency-division multiplexed into a single antenna.
6. At least three antennas required per terminal.

5. RECOMMENDATIONS FOR FUTURE WORK

Many of the techniques peculiar to a passive, spherical, satellite communications network will require experimental verification before they can be incorporated into an actual system. A large number of these are concerned with such factors as orbital perturbation, antenna design, tracking, modulation, and anti-jamming. Relatively few are in the terminal equipment area. This is due to the fact that many proven techniques can be utilized in this application.

The principal novelties of the terminal equipment proposed in this report, as compared to the terminal equipment for a conventional system, are the time-division multiplexing of all types of traffic for a link into a single bit stream, the periodic interruption of store-and-forward traffic to provide satellite transfer intervals, and the reclocking of received bit streams to compensate for Doppler effect and asynchronism between station clocks.

Although relatively unique as recommended for the reference network, time-division multiplexing is quite common in other applications. If necessary, tests could be performed in this field without

the use of satellites to verify the feasibility of combining four different types of traffic into a single bit stream.

The transfer of the communication path from one satellite to another and the reclocking of the bit stream will best be verified by experimental tests with actual satellites. Since at least two satellites are required for this purpose, these tests could not be performed until after the multiple launch of three passive spheres by NASA scheduled for 1963. Pattern generators could be used to simulate teletype and data traffic, but in the case of voice and graphics, actual traffic will prove more useful. Some points to be verified by these experiments are the length of the transfer interval required, the speed and ease of re-establishing synchronization after a transfer, the effect of satellite transfer on voice and graphics information, and the effect on voice traffic of the delay introduced by reclocking.

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APPENDIX I

SOME CONSIDERATIONS FOR A FUTURE GROUND-SATELLITE-GROUND COMMUNICATION SYSTEM

Introduction

A method for communicating ground-satellite-ground has been proposed, using wired logic at the terminal points. This method used conventional techniques and is the electronic equivalent of a present-day electromechanical switching center modified to accomodate the transmission problems unique to satellite systems.

It is the purpose of this section to introduce more sophisticated concepts, which while requiring some development, would reduce the amount of hardware used, and simultaneously increase the utility of the system and its capacity. In addition, various other functions that are not strictly switching functions could be accomplished.

The new concept is one using a stored program, digital computer as a central switching control at terminals with special hardware to facilitate the rapid input and output of data through various lines into and out of the computer. This concept is the one being implemented for 465L and ComLogNet, and seems well suited for the ultimate satellite net terminals.

The presently proposed equipment uses wired logic, essentially on a per line basis. As an example, there is one tape machine of special design (two heads, one for reading and one for writing, plus mechanisms to guard against tight-tape conditions) per precedence, per line. In addition, there are per line buffer stores. The system outlined here reduces the per line storage, and no tape machines are used. All functions are accomplished in a central memory.

The use of a stored program computer rather than a wired logic scheme would make it possible to use the periods when a satellite illuminates three ground stations. This accrues from the adaptability of a stored program machine, to accomplish several functions using the same

hardware, and making rapid calculation of when to accomplish each function.

The stored program machine can be used to accomplish the sending of store-and-forward messages directly from station j to station $j+2$, thus bypassing a tandem station. Routing line segregation can be accomplished by the computer so that this very short time can be utilized to send S/F messages on all trunks during the interval. This saves tandem storage and relieves S/F backlogs.

The system using stored program shares trunks so that S/F and direct traffic are handled by the same equipment and the same trunks. This makes trunk use much more efficient and avoids the use of two separate networks.

The wired logic network does not provide continuous service between adjacent "A" stations. One reason for this is the difficulty of calculating satellite data accurately, and using this to compensate exactly for the difference in delay incurred when switching from one satellite to another. Stored program logic makes possible complete synchronization between "A" stations.

Basic Considerations

Each ground station complex contains a stored program computer with a central high-speed memory (core, thin film or some other rapid store). Associated with this, is a disc file used to store waiting (store-and-forward) messages.

The system is digital, and analog signals, such as voice, must be converted to a digital signal by appropriate means before entering the system. This is accomplished by analog to digital converters at the first ground station where the local subscriber originated signal enters the system.

All satellite trunks are time division switched, but signals from local subscribers entering a station are space division switched into

the ground station.

A directed transmission link is defined by a particular carrier frequency; the return link uses a different frequency. Thus, both ends of a transmission link and the direction of transmission are uniquely determined by the carrier frequency.

Each carrier is time divided so that a number of channels (trunks) are provided by each transmission path. Each trunk can accommodate the equivalent of one PCM voice channel (at the appropriate rate required for this service.) Lower speed S/F traffic is normalized (speed buffered) to this speed.

Trunks are not permanently assigned to any service but are shared. This is possible because all transmissions are at the same speed. A trunk may handle direct switched traffic at one time and S/F traffic at another time. The only permanently assigned trunks are, one (or more) for supervisory information and one for synchronizing pulses.

Rather than an originator determining whether a message is to be direct switched or sent store-and-forward, the mode of transmission (direct switched or S/F) will be determined by the type of message. Thus, all teletype and data will be sent S/F, while voice and such messages as FAX will be directly switched. (Direct switching is necessary in the case of FAX, to avoid using the very large storage required to hold a FAX message if it were sent S/F.)

Direct switched messages can be sent only in real time. For S/F traffic, messages could be broken up, or could be sent out faster, by spreading them over a number of trunks. However, this would require more complex equipment, and no advantage seems to accrue from either procedure. Thus, S/F messages will be sent complete on one trunk. The trunks are utilized just as efficiently, because if more than one trunk becomes available for S/F messages, then more than one S/F message can be sent simultaneously, one per trunk.

As described later, transmissions between "A" stations may not require resynchronization when switching from one satellite to another. It will however, be necessary to resynchronize between an "A" and a "B" station after each interruption of service. This will be accomplished in the usual fashion by sending out a given pattern that is recognized by the receiving station in the synchronizing pulse time slot. Two methods can be used for recognizing the signal pattern. One method is to successively search each of the time slots until the pattern is recognized. This method requires less equipment, but takes more time. The second method is one in which the bits in the various time slots of the bit stream are observed simultaneously. This method utilizes a delay line. The incoming bit stream is sent into a delay line, and the various time slots are looked at simultaneously at taps on the line. A pattern is recognized more quickly thus allowing more rapid synchronization than the first method.

Ground Station Equipment and Operation

Each receiving station maintains a separate oscillator plus an associated counter, which counts up to the number of channels in a carrier, for each incoming carrier. This is used to demultiplex the channels. (The synchronizing pulses synchronize the oscillator, and the counter counts oscillator pulses, thus separating out the various time slots.)

Each station has a computer with a certain number of ingoing access routes, which in general, is less than the total number of incoming channels from all received carriers. Each ingoing channel to the computer has associated with it two bits of storage, one to store the information bit and one to store the clock bit. (The presence of a clock bit indicates that there is information present.) One ingoing channel into the computer is permanently assigned to each supervisory incoming channel. Other channels are assigned on the basis of supervisory information received.

Each ground station has an equipment similar to a telephone local office to handle local subscribers. A local subscriber wishing to

send a message, sends the equivalent of an off-hook signal to the local office. The local office assigns a register to the subscriber (if one is available) and sends the equivalent of dial tone to the subscriber (or nothing, if no register is available). The dial tone is taken as a go-ahead signal by the subscriber. (The number of registers is a certain percentage of the total number of subscribers. A register consists of one bit of storage for information plus one bit of storage for the equivalent of clock.) A register receives one bit of information. The various registers are scanned at a rate equal to the local computer clock rate. This is high enough so that only one new bit can arrive at a given register between scans. (Registers assigned to higher speed lines are scanned more often.) The output is at the clock rate and goes to the computer as if it were one of the channels from a satellite link. The order of scanning is stored by the computer so that it can separate the various inputs. (If the combined bit rate is more than can be handled by a single channel, more ingoing channels to the computer can be used.

The computer is time shared among its various functions. Among these functions is the assembly of headings (or their equivalent) and acting on them.

The computer uses a disc file as storage waiting queues of messages. It is not proposed to use any other backup store. A disc will provide storage for about 2×10^8 bits, with an access time of 30 milliseconds. Thus, a large number of messages can be stored; considering what data is presently available, this seems adequate. If it becomes necessary, a further backup store using a tape machine can be implemented.

The use of a disc as backup store enhances the efficiency of the computer in that less rapid access memory will be required to provide buffering between lines and disc than, for example, would be required between lines and a tape machine. This is due to the fast access time for a disc. Of course, a drum could be used as intermediate buffer between lines and tape; but this would be double buffering, hence, more complex

and expensive. It is proposed that messages be assembled completely in rapid access memory before being transferred to the disc file. This makes bookkeeping easier. Similarly, a complex message at a time would be transferred from disc to central store.

Supervisory Signalling

Now, let us consider how direct switched and S/F calls are set up, and how the calls are handled.

For direct switched calls, a subscriber goes off-hook, and a register is assigned. He dials his heading into the computer, whereupon the computer reads the heading, assigns an incoming time slot to the subscriber, and a free outgoing time slot to this, and then signals on the supervisory channel to the next tandem point. At this point, the computer reads the supervisory message, assigns time slot, and signals ahead. This process continues until the destination ground station is reached. The destination ground station then activates the ringing signal to the distant subscriber and sends a signal via the return supervisory channel to the originating ground station (via tandem points which relay the message). The ground station then sends a signal indicating that the distant subscriber is being called. When the distant subscriber goes off-hook (or equivalent), the sender begins his message. For example, if it is voice, a digital encoder is inserted in the line, and voice signals are changed to digital signals going into the computer. These are then sent over the assigned time slots, reconverted at the receiving end, and sent to the local subscriber converter at that end. Because of high computer speeds, the whole setting up of the call is done very rapidly. For conference calls, the process is similar except that more destinées must be reached.

For store-and-forward calls, the procedure is somewhat simpler. When a store-and-forward heading is received by the computer, a space in memory is assigned by the computer, and a go ahead sent to the local subscriber. The message is received by the computer, assembled and put in an appropriate queue waiting to be sent to the next tandem point. (Actually the message will be written in a random location in the backup store, which is a magnetic disc file).

When a channel is available, the computer signals on the supervisory channel; upon receiving an acknowledgment, it begins sending the S/F message, which then enters storage at the tandem point. This process continues until the destinee is reached. In the case of a multiple address, the message may be sent out many times in various directions.

In any of the cases previously mentioned, if time slots are not available, a busy signal is returned. In the case of high precedence calls, a computer may pre-empt a time slot, sending a cancel or busy signal to the interrupted subscriber, and use the time slot for a higher precedence call.

Supervisory messages are used mainly to indicate which message to send and which connections to make; however, supervisory messages will also be used to exchange various traffic data among stations and to signal stations, if an outage condition exists. Basically, there two types of supervisory message, the request for connection message, and the general information message.

A request for connection message has the following structure:

- (a) Beginning of message symbol.
- (b) Symbol indicating type of supervisory message (in this case Request for Connection).
- (c) Code number indicating type of message referred to (teletype, voice, etc.) (This also indicates whether the message is direct or S/F.).
- (d) Type of connection requested. (Single direct connection, conference call, etc.).
- (e) Addressee (s) code number (s).
- (f) End of message symbol.

While a general information message has the following structure:

- (a) Beginning of message symbol.
- (b) Type of message symbol (General Information).
- (c) Text of message.

- (d) End of text symbol.
- (e) End of message symbol.

Alternate Routing

Inasmuch as the proposed system utilizes a single string of "A" stations which is not a complete belt, alternate routing, in the usual sense, must, of necessity, be extremely limited. It is possible to use "B" stations to perform some alternate routing tasks; however, these "B" stations, cannot normally be used as alternates for "A" stations because illumination of one "B" station and one "A" station occurs only for short intervals. In addition, "B" stations are not usually geographically located so that there is any long interval during which a "B" station, and an "A" station which is not its local "A" station, are simultaneously illuminated by a satellite. Also, "B" stations generally have less trunks to switch than "A" stations and will probably have less sophisticated terminal equipment, thus severely limiting the volume of traffic that can be handled.

In view of the above arguments, it is not proposed to use any form of alternate routing to equalize traffic loads. However, alternate routing, in the sense of graceful degradation of the system, should be provided. This is especially desirable to allow the system to give some limited service during outages due to enemy attack or any other reason. The following scheme of operation during outage conditions is proposed. If an outage occurs at any given "A" station, the closest "B" station will send out a supervisory message (if the "A" station is completely disabled). (Of course, this message must wait for the proper satellite illumination condition.) All direct switched traffic that must go through the disabled "A" station will cease. In addition, routine S/F traffic will not be sent. Only high precedence S/F messages that must pass through this point will be accepted by other stations.

System Analysis

Mixing of S/F and Direct Switched Traffic on Trunks - This section discusses an approximate analysis of delays that store-and-forward traffic would

encounter when mixed on the same trunks with direct traffic. For the purpose of this appendix, it is assumed that the only delays encountered by S/F traffic are those incurred in waiting for a trunk to be assigned. This analysis is for delays encountered between two adjacent node points of the network.

Inasmuch as no data is available as to maximum allowable delays, it is not possible to specify a system of assigning of trunks to store-and-forward service. However, in general, it is neither practical nor possible to design trunk groups to definite delay limits without an attendant probability of such delay. To make use of probability calculations and obtain a reasonable degree of economy, a certain percentage of the offered traffic load must be allowed to exceed these time limits. If this percentage can be kept small enough, and the probability of occurrence of intolerable delay still smaller, it should be possible to design a trunk group to handle direct traffic and meet delay limits for indirect traffic to some very high degree. Different precedence levels are not considered separately. Higher precedence messages are serviced first, but all S/F messages are treated similarly as far as trunk access is concerned.

Small Amount of S/F Traffic - The following calculations are based on a Poisson distributed called incidence.

It is assumed that direct traffic is characterized by constant holding time (HT). This will cause results in terms of delays for S/F traffic which are too optimistic as compared with a more realistic exponential distribution of the same average. Moreover, it is assumed that if an S/F message is unsuccessful, the next attempt to send the message will be one HT later. The purpose of the above assumption is to make each attempt independent of previous attempts. After one HT has elapsed, no correlation exists with the previous situation of trunk occupancy. Thus, the S/F message unsuccessful on the first attempt will experience the same

probability (P_0) of being served on the second attempt. This fraction of S/F traffic $(1 - P_0)$ will make a second attempt, thus $(1 - P_0) P_0$ will be served on the second attempt, experiencing a delay of one HT, and $(1 - P_0)^2$ will be unsuccessful on the second attempt, experiencing a delay of two HT or more. In general, the probability distribution for experiencing a delay of Z holding time (ZT) is given by

$$P(ZT) = (1 - P_0)^Z P_0$$

for $Z = 0, 1, 2, \dots$

The average delay on all S/F calls can be calculated as follows -

$$\begin{aligned} P_{av} &= \sum_{Z=0}^{\infty} P(ZT) \cdot ZT = P_0 T \sum_{Z=1}^{\infty} Z(1 - P_0)^Z \\ &= P_0 T \frac{(1 - P_0)}{P_0^2} = T \frac{(1 - P_0)}{P_0} \end{aligned}$$

where

$$\begin{aligned} \sum_{Z=1}^{\infty} Z(1 - P_0)^Z &= (1 - P_0) + 2(1 - P_0)^2 + 3(1 - P_0)^3 + \dots \\ &= (1 - P_0) + (1 - P_0)^2 + (1 - P_0)^3 \\ &\quad + (1 - P_0)^2 + (1 - P_0)^3 \\ &\quad + (1 - P_0)^3 \\ &\quad + \dots \\ &= \sum_{n=0}^{\infty} \sum_{Z=1}^{\infty} (1 - P_0)^{Z+n} = \sum_{n=0}^{\infty} \frac{(1 - P_0)^n (1 - P_0)}{P_0} \\ &= \frac{(1 - P_0)}{P_0^2} \end{aligned}$$

Appreciable S/F traffic - As before, no specific distribution of HT's of S/F traffic is assumed. However, the assumption is made that the average HT of an S/F message is short, compared with the average HT of direct traffic, so that flow of direct traffic can be regarded as effectively not impeded by the S/F traffic.

The probability of zero trunk occupancy when S/F traffic is sent in an hour is given by P_{0-b} , where P_0 is, as before, the percentage of zero trunk occupancy when only direct traffic is sent; and b is the percentage occupancy of S/F traffic. Thus, a larger proportion of S/F messages will experience delay. As previously noted, an S/F message unsuccessful on an attempt will try again one HT later. (This, as before, leads to independence between the n^{th} and $(n-1)^{\text{st}}$ try). Then similarly to the above equation, the distribution of delays is given by -

$$P(ZT) = [1 - P_0(b)]^Z \quad P_0(b) = (1 - P_0 + b)^Z (P_0 + b)$$

Two series links - Suppose we want to find the probability of a given delay for a path which traverses two links. We can approximate a result, by taking finite step probabilities. For example, consider two discreet probabilities for link one:

P_{11} that delay is less than ρ_1

and P_{12} that delay is greater than ρ_1

similarly for link 2:

We have P_{21} that delay is less than ρ_2

P_{22} that delay is greater than ρ_2

We approximate the probability of getting a delay $\rho = \rho_1 + \rho_2$ by taking $P_{11} P_{22} + P_{12} P_{21}$. This result is a very crude approximation. Now consider more than two possible time intervals and attendant probabilities. For link 1 we have: P_{11} that $\delta_1 < \rho_1$,

$$\begin{aligned} &P_{12} \text{ that } \rho_{11} \leq \delta_1 < \rho_{12} \\ &\vdots \\ &P_{1n} \text{ that } \rho_{n-1} \leq \delta_1 < \rho_{1n} \\ &P_{1n+1} \text{ that } \rho_{1n} \leq \delta_1 \end{aligned}$$

Similarly for link 2 we have:

$$\begin{aligned} &P_{21} \text{ that } \delta_2 < \rho_{21} \\ &\vdots \\ &P_{2n} \text{ that } \rho_{n-1} \leq \delta_2 < \rho_n \\ &P_{2n+1} \text{ that } \rho_{2n} \leq \delta_2 \end{aligned}$$

If the number n is large, the approximation is much better. If we let $n \rightarrow \infty$ so that the intervals approach zero, then we have for the limit

$$P = \sum_{j=1}^{\infty} \sum_{i=1}^{\infty} P_i P_j$$

For n links

$$P = \sum_{j=1}^{\infty} \sum_{i=1}^{\infty} \cdots \sum_{R=1}^{\infty} P_i P_j \cdots P_R$$

The terms P_i must each be found by the methods already given.

We can approximate the probabilities as well as we want, by making n as large as we want. A computer program to find the above results is easy to set up, and will yield the required results.

APPENDIX II

SINGLE BIT STREAM/LINK VS TWO BIT STREAMS/LINK

In the body of this report it is proposed that a single bit stream be used for each simplex link. An alternative to this scheme is the use of two independent bit streams; a high speed stream for real-time traffic and a lower speed stream for store-and-forward traffic. A separate subcarrier could be used for each stream.

The functional diagrams in figures 11 and 12 can be used for two bit streams per link if slight modifications are made. The store-and-forward channels shown leaving the demultiplexers and entering the multiplexers in figure 11 should be disregarded. In figure 12 a multiplexer and a demultiplexer must be added to each link to perform the functions formerly performed by the corresponding equipment in the real-time center and a reclocking circuit must be added to each incoming stream.

As with the single bit stream, the speeds of the two bit streams will depend upon the number of channels per link. A comparison of the relative bit rates involved in the two concepts may be obtained by using the same bit rates per channel as was specified in the body of this report for each type of traffic; i.e., 38,400 bits per second for voice, 9,600 bits per second for graphics, 2,400 bits per second for data, and 75 bits per second for teletype. The resulting stream bit rates are:

$$\begin{aligned}\text{Real-time stream} &= \text{Voice channel bit rate (voice channels + 3)} \\ &= 38,400 (\text{voice channels} + 3)\end{aligned}$$

The bit rate for six graphics channels with phase

indicator bits is two times the voice channel rate or 76,800 bits/sec. Additional graphics channels in multiples of six can be added by an increase in the bit stream of 76,800 bits/sec. A third voice channel (38,400 bits/sec) is added for control purposes. This latter channel provides framing and sync information as well as an indication of simulated traffic.

Store-and-forward Stream = Teletype channel bit rate (teletype channels)
+ Data channel bit rate (data channels)
+ Control channel bit rate
= 75 (TTY channels) + 2400 (data channels)
+ Control channel bit rate

As in the case of the single bit stream per link, the control channel provides framing and sync information, retransmission requests, simulated traffic indication, and start of transfer intervals (if used). This channel should consist of 3 or 4 bits during each framing interval. With an 8-bit code such as Fieldata, the addition of the control channel will result in a 50 per cent increase in the store-and-forward bit stream rate.

The two separate bit streams per link are thus seen to result in the duplication of several functions for each stream. These include multiplexing, reclocking, demultiplexing, and control channel functions.

The real-time bit stream is at the same speed as the single combined bit stream. The store-and-forward information has only been replaced by 3 graphics channels. Therefore, the single bit stream is considered to be more efficient and, hence, preferable.

APPENDIX III

OPTIMUM PCM TECHNIQUES FOR REFERENCE SYSTEM

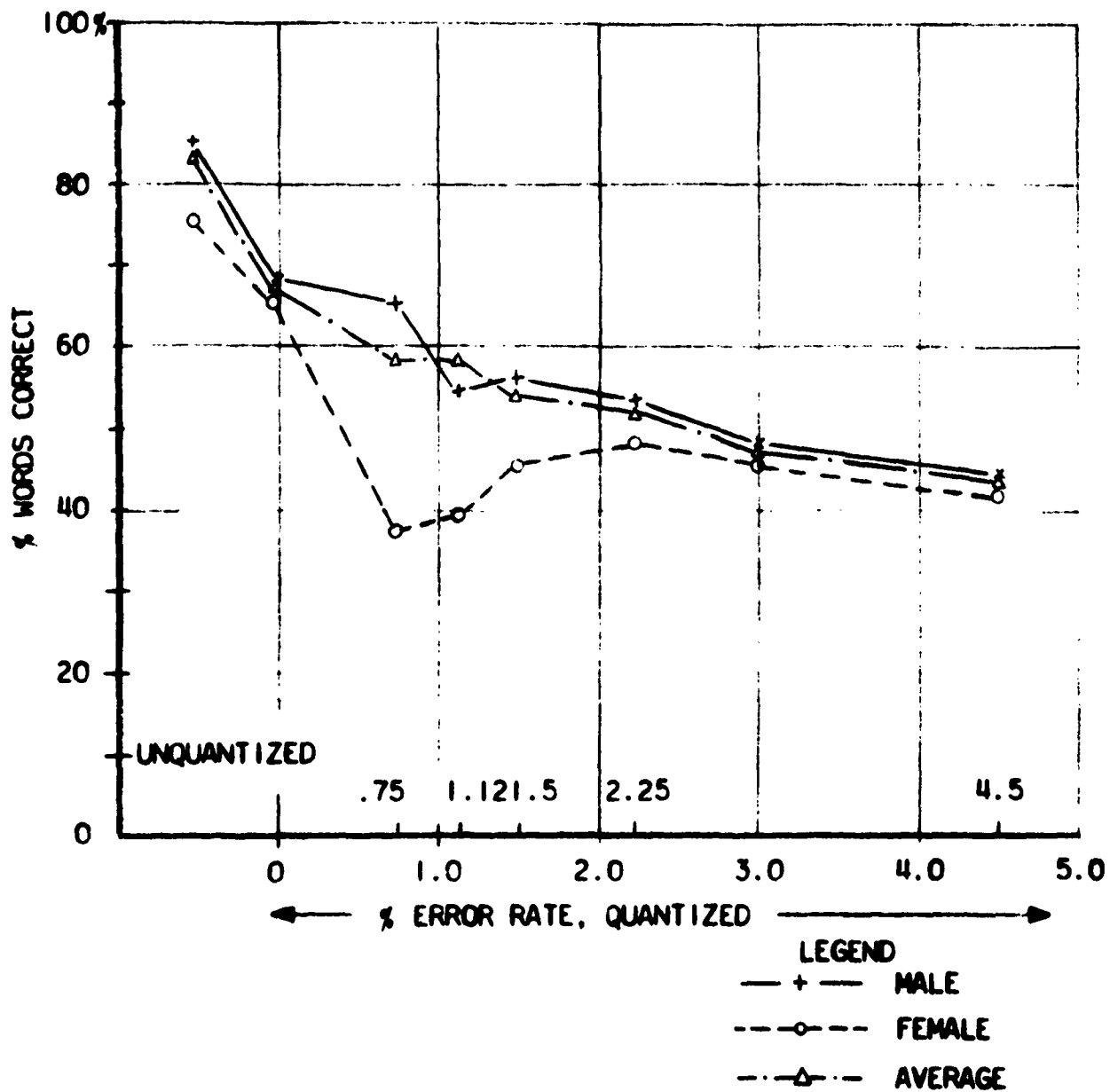
Pulse code modulation (PCM) has been recommended for the transmission of digital signals in the reference network. This selection was based upon a number of factors. On the negative side, PCM requires more bandwidth than the direct transmission of the signal itself. For instance, to send a signal of bandwidth W_0 by PCM requires $2W_0$ code groups per second with each code group containing n bits. This results in a bit rate of $2nW_0$ pulses per second which in turn requires a bandwidth $W = nW_0^{135}$. Several advantages are obtained in exchange for the extra bandwidth. Since each pulse is transmitted as either a "one" or a "zero," there must be an in-between level above which a pulse is considered to be a "one" and below which it is considered to be a "zero." If the noise exceeds this "threshold" level, an error results. Therefore, there is a fairly definite signal-to-noise ratio of about 20 db, below which the interference is serious and above which the interference is negligible. Comparing this figure with the 60 to 70 db required for high quality am. transmission of speech, it is obvious that PCM requires much less signal power. A high quality PCM signal can be obtained under extremely poor conditions of noise and interference because it is only necessary to recognize the presence or absence of each pulse. In addition, by using regenerative repeaters which detect the presence or absence of pulses and then emit reshaped, respaced pulses, the initial signal-to-noise ratio can be maintained through a long chain of repeaters. PCM also lends itself well to time-division multiplexing and secure encryption. It was considered that the advantages of PCM far outweighed its additional bandwidth.

Since PCM requires $2W_0$ code groups per second to send a signal of bandwidth W_0^{135} , this amounts to 8000 code groups per second for a normal speech band of 4000 cycles. The resulting bit rate depends upon the number of bits per code group, which in turn, depends upon the quality of speech required. Quantizing noise results from assigning a continuous range of levels to several discrete steps. The fewer the steps (or combination of bits) the greater the noise. Tests at ITT Federal Laboratories show the following quantization noise ratios for various number of coding bits:

QUANTIZATION NOISE RATIOS FOR VARIOUS NUMBER OF CODING BITS²

<u>PCM Coding Bits</u>	<u>Quantization Noise Ratio in db</u>
1	8
2	14
3	20
4	26
5	32
6	39
7	45
8	52
9	58

A compromise must be arrived at between a small number of coding bits to save bandwidth and a greater number to increase intelligibility. Speech intelligibility is usually measured by the percentage of persons in a test group who can understand the speech. Recent tests at ITT Laboratories produced the following results:



Vocoder Intelligibility Scores
 1500 bits/second 8 Channels

<u>Bit Rate</u>	<u>Modulation</u>	<u>Low Pass Filter</u>	<u>% Intelligibility</u>
16 Kb	4 bit PCM	1800 cps	83
19.2 Kb	3 bit PCM	3100 cps	84
38.4 Kb	4 bit PCM	3500 cps	92
38.4 Kb	6 bit PCM	3100 cps	95
48 Kb	6 bit PCM	3500 cps	97

Differential PCM encodes the difference in amplitude between successive speech samples instead of the actual amplitude. Since each change in amplitude is considerably smaller than the total amplitude, Delta PCM can more accurately represent the speech waveform by the use of smaller discrete levels. ITT intelligibility tests with Delta PCM are summarized below:

<u>Bit Rate</u>	<u>Modulation</u>	<u>Low Pass Filter</u>	<u>% Intelligibility</u>
38.4 Kb	4 bit Delta PCM	3500	96
38.4 Kb	6 bit Delta PCM	3100	96
48 Kb	6 bit Delta PCM	3500	98

Other experiments³ indicate that for normal speech, four-digit differential PCM is comparable to six-digit normal PCM.

Speech bandwidth compression devices such as low bit rate vocoders (2400 - 4800 bits per second) do not have extremely high intelligibility and suffer badly when there are errors in transmission. Higher bit rate vocoders (9600 bits per second) are still being developed with the objective of obtaining toll quality.²

Since toll quality speech with speaker recognition is desired by the Air Force, 4-bit differential PCM resulting in an excellent quality voice channel of 38,400 bits per second is proposed for use in the reference network.

Four-bit differential PCM also offers a possibility for use with digitized graphics. Experimental work³ indicates that a four-digit coder of the type recommended for speech transmission transmits a picture of comparable quality to a six-digit standard PCM coder if no errors are involved in the transmission. Unfortunately, a transmission error can shift the absolute value of the recovered signal and this can remain uncorrected long enough to be objectionable in a picture. This type of error is not important in the speech case and can be gradually corrected.

Because of the problems in the use of 4-bit differential PCM for graphics, an equivalent 6-bit normal PCM is recommended for use in the reference network. In the present state of the art, this is the minimum code necessary to produce a picture with barely detectable distortion when reproduced. With a sampling of the facsimile baseband signal at a rate of 1,600 times per second, the resulting PCM signal will contain 9,600 bits per second. Faster sampling would produce correspondingly faster bit rates.

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APPENDIX IV

ERROR DETECTION AND CORRECTION

With long-distance radio transmission of high-speed digital data, use of some type of error-detecting or error-correcting code is desirable to insure the accuracy of the received information. This insured accuracy has the disadvantage of an increase in the bandwidth. The simplest error-detecting scheme is the addition of a single parity bit to each binary character. This bit is used to make the total number of "1" bits in the character an odd or even number depending upon whether odd or even parity is used. If the received signal fails to contain all odd bit or all even bit characters, an error is indicated. This difficulty can be eliminated by separating the digital transmission into blocks and checking parity in both the horizontal and vertical directions. This detects the error but does not correct it. In order to correct the error, additional parity bits must be added to each character, or the character which was incorrectly received must be retransmitted. The latter method is recommended for use in the reference network for the following reasons:

1. Error-correcting codes require a large number of parity bits (and consequent bandwidth) as shown by the expression 1 for single correction:

$$2^p \quad m + p + 1$$

where p = number of parity bits and m = number of information bits. This indicates that for a six-bit information code, four parity bits are needed in each character for single error bit correction.

2. The electronic circuitry required for error correction is more complex and costly than that needed for error detection.
3. If more bits are in error than the error-correcting code is designed to handle or an entire character is lost in transmission, an error-correcting code is unable to provide the necessary information.

To permit error correction by retransmission, a return communications path must be provided. This is made available by the use of certain bits in the control channel of the return bit stream in the same link. Duplex operation is already required on nearly all links because of the use of voice channels. If simplex store-and-forward operation should prove necessary because of jamming or equipment outage, the error-correcting feature could be dispensed with and all messages accepted as received or hamming type codes used.

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APPENDIX V

PROBLEMS ARISING FROM A VARIABLE TIME DELAY INSERTED IN A TRANSMISSION PATH

A satellite transmission system can be represented as a four terminal network separating input and output. This network is neither linear nor time invariant; hence, special attention must be given to these factors in the design of terminal equipment.

The system can be specified schematically as the combination of a conventional linear, time invariant element $H(w)$, a nonlinear term $H(a)$, and a time dependent term $H(t)$. In the most general case these terms are inter-related and should be expressed as $H(w, a, t)$; in the real systems under consideration, certain simplifying assumptions can be made.

- (a) $H(a) = Ka_{in}$, that is the system is linear or can be made sufficiently so for practical consideration since the linearities arise primarily in the ground equipment.
- (b) $H(w) = \text{flat and real}$ or at least characterized by linear phase shift. If this is not true for modulation techniques requiring sideband coherence over a wide spectrum, then piecewise transmission in a series of narrow bands is certainly possible without excessive distortion of the baseband intelligence.
- (c) $H(t)$ in a satellite communication system represents a variable time delay arising from the continuously

varying distance from station to satellite to station. Its effect is apparent in the carrier as a doppler and in the modulation as a corresponding change in bit rate or modulation frequency. In this case, the remedies applied to the radio transmission path may not remedy the information rate problem.

The solution to the time delay problem can be considered in the following form where $H(t)$ is a pure but variable time delay $S_t = H(t)$

$$e(t) \quad H(t) \quad G(t) \quad e_2(t + \Delta t)$$

To make $e(t)$ and $e_2(t + \Delta t)$ identical in form, $H(t)$ and $G(t)$ must be compensating functions. Since truly anticipatory circuits are impossible, $G(t)$ must in reality be always positive, hence

$$G(t) = \Delta t - H(t)$$

where

$$\Delta t = H(t)_{\max} - H(t)_{\min}$$

It follows therefore that the net transfer delay is

$$H(t) + G(t) = H(t) - \Delta t + H(t) = \Delta t$$

and even with optimum time compensation, a time delay of Δt will be added to the already existing path delays. In physical terms this is equivalent to building out the delay of the link by a variable delay line so that the total delay of transmission path plus variable delay line is equal at all times to the longest delay normally anticipated in the path. The variation in delay necessary is $H(t)_{\max} - H(t)_{\min}$.

There are many ways of accomplishing this delay, ranging from a motor driven tapped delay line to memory devices that are loaded and unloaded at different rates. The applicability of the various techniques

depends to a large degree on the magnitude and rate of change of delays anticipated in the particular system. These are considered in the following manner:

Maximum Delay - In a satellite system the maximum possible delay occurs when both stations view the satellite simultaneously at maximum range. The time delay in transmission at this point is $1/2 H(t) = \frac{R_{\max.}}{c}$ and is independent of station spacing.

Minimum Delay - The minimum range is in turn a function of orbit altitude and station spacing. The smallest value of minimum range will occur when the communication stations are adjacent to one another and the satellite is overhead. This is the limiting case and will be used in establishing minimum time delay.

For any orbit altitude the maximum time delay difference is easily obtained by taking the difference of these two values. For example, at an altitude of 2,200 nautical miles the maximum delay difference assuming a 5 degree antenna elevation at maximum range will be 25-13 or 12 milliseconds for each path or $2 \times 12 = 24$ milliseconds total. It is interesting to note that this difference remains essentially constant with increasing altitude as evidenced in the following table.

Transmission Delay Differences (for 5 degree antenna elevation at max. range)

<u>Orbit Altitude (nmi.)</u>	<u>Δt (ms)</u>
900	19
1500	21
2100	24
2700	25.5
3900	27.5
5100	30

In short, the techniques applicable in time delay corrections at 2,200-nmi. altitude are in fact applicable to all orbits of interest for spherical passive reflectors.

Applicable Techniques

Variable Delay Line - A delay line whose electrical length is varied either mechanically or electrically could be inserted in the RF path to accomplish both carrier and modulation delay. At RF, this would require an unreasonable length of line (in the order of 10^6 feet); hence, it is not further considered.

At modulation frequencies, however, various delay devices exist which might be used. Distributed delay lines having mc bandwidths and time delays of 1 μ sec/ft are available but again would require lengths in the order of 10^4 feet. Lumped constant delay lines are characterized by delays per section in the order of the reciprocal of their bandwidth. In digital systems this implies a delay of one bit per section. Here a delay line of possibly 23,000 sections might accomplish the required maximum delay. The line loss, however, and the necessary commutation to achieve the delay variation make this approach prohibitively complex.

Digital Delay Devices - The time delay problem fortunately is analogous to the speed buffering problem in digital transmission and switching techniques. In this case, two tools are of particular interest; i.e., the looped magnetic tape with independent, variable-speed read and write capstans; and the shift register.

The magnetic tape is uniquely suited to speed conversion. The input or record drive operates at a constant speed and records all input data in real (though variably delayed) time. The output or read capstan,

however, is operated at a variable speed so that the output bit stream is synchronous with the local station clock. The output speed, therefore, is determined from a feedback loop which operates on the phase error between readout pulse train and local clock. This appears an optimum variable delay; however, because it is mechanical in nature, it possesses disadvantages arising from its inability to change speeds instantaneously. While it can easily follow doppler changes once synchronized, it requires considerable time to establish this synchronization. (Minimum stop/start times for the Ampex FR600, operating at 38 in/sec., is in the order of 50 to 100 ms.).

While this may not be unreasonable for stop/start operation, a problem arises during transfer from one satellite to another. Even when this takes place at equal delay, so that the information stream is uninterrupted, it is theoretically impossible to accomplish this transfer without an abrupt change in doppler. This follows logically from the fact that the "setting" satellite is characterized by a negative doppler while the "rising" satellite is characterized by a positive doppler. For a satellite at a 2,200 nmi. orbit, this change could be as much as 26 parts per million. The tape machine would have to make this correction before a bit was lost at the new rate. It would probably be necessary for reclocking to take place before a time shift of approximately 1/8 bit was accumulated. For a 50-kb pulse train, for example, this will occur in about

$$T = \frac{1}{8} \times \frac{1}{50 \times 10^3} \times \frac{10^6}{26} = 0.1 \text{ sec.}$$

The situation can be somewhat simplified if no useful information is transmitted during transfer; however, the interruptions required would be in the order of a $1/4$ second which could not be tolerated.

This effect does depend upon orbit altitude and becomes progressively worse as the altitude decreases.

Buffer Storage-Speed Conversion - A technique analogous to the tape memory but possessing a considerably shorter reaction time is the shift register buffer. In this technique the digital data is read into a shift register in real time. When the shift register capacity is reached, the input pulse train is automatically switched to a second register and the first one is read out in synchronism with the local clock. Any changes in input pulse rate are accommodated within the interval of at most a few pulses.

The capacity of the buffer required for transmission delay is easily calculable from

$$N = 2 \ txf$$

where

N = number of bits of storage

t = maximum difference in time delay

f = average input bit rate

The factor of 2 arises from the uncertainty as to whether the initial conditions at read-in represent minimum or maximum delay. In this case, when the register is first loaded, readout is delayed for approximately one half the total delay. If doppler prediction, or more accurately range sum prediction, data is available, this factor can be reduced toward one, as a function of the accuracy of prediction.

In the example under discussion (e.g., 50 kb 25 ms delay) the shift register (for the worst condition) would require a capacity of

$$N = 2 \times 25 \text{ ms} \times 50 \text{ kb} = 2500 \text{ bits}$$

which most likely would be achieved in a simple core matrix (50 x 50).

An additional factor must be considered, namely, the accuracy of the local clock. The total transmission time delay cannot increase without limit, and consequently a finite limit exists for buffer store capacity. If the clocks at the various stations are not in synchronism, a continuous excess or deficit of pulses develops and the store capacities are exceeded. The additional store requirements are

$$N_e = \int_0^T f \times dt$$

where

N_e = excess capacity required

Δf = clock error (cps at bit rate)

T = interval of observation

To cite an example, a local clock having a stability of 1×10^{-8} (referenced to WWV or WWVH) would, if running at maximum tolerance for one week without correction, require an excess buffer capacity of

$$\begin{aligned} N_e &= 50 \times 10^3 \times 1 \times 10^{-8} \times 3600 \times 24 \times 7 \\ &= 300 \text{ bits} \end{aligned}$$

This is certainly not an excessive requirement.

It is necessary, however, that once each week the station clocks be reset. It is possible that a measure of frequency error relative to all of the adjacent stations can be achieved by sampling the buffer capacity and averaging. By systematically starting from the gateway station (one only) and proceeding outward on the link to successive A stations, then to

their associated B stations, and finally to the C stations a system calibration can be affected. This calibration can be done while handling traffic; therefore, it should not compromise the operating reliability of the system.